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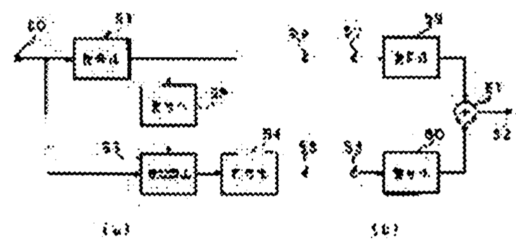
(54) HIGH EFFICIENCY CODING AND DECODING SYSTEM

(57)Abstract:

PURPOSE: To prevent generation of a pre-echo in the system where interchangeability is provided even to a reproduction device fixed by an existing low bit rate, the system with high sound quality using a higher bit rate is able to be introduced and bits are arranged completely optimizingly with respect to signals of every property.



CONSTITUTION: A high efficiency coder is provided with plural coding circuits 51, 54, a decoding circuit 52 decoding a signal subjected to coding processing, and a difference calculation circuit 53 calculating a difference between an input signal and a signal subjected to decoding processing. Then the input signal is coded and the coded signal is decoded and a difference between the decoded signal and the input signal is further coded and the result is transmitted to the high efficiency decoder together with the coded input signal. Furthermore, the high efficiency decoder is provided with plural decoding circuits 59, 60 and an adder circuit 61 synthesizing decoded signals and the coded signal from the high efficiency coder is decoded and the result is synthesized on time base and output signal is obtained.

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DETAILED DESCRIPTION

[Detailed Description of the Invention]
[0001]
[Industrial Application] This invention relates to high efficiency coding and the decryption system which consist of the high-efficiency-coding equipment which reduces the bit rates used in stereos, such as a digital audio signal recorded message sender for telephone and a motion-picture film projection system, or the so-called multi-surround sound system, a transmission medium with which the signal encoded by this equipment is transmitted or recorded, and high efficiency decryption equipment which decrypts the signal transmitted and encoded.
[0002]
[Description of the Prior Art] Although it is in the technique and the equipments of high efficiency coding of a signal, such as an audio or voice, variously For example, block the audio signal of a time domain etc. for every unit time amount, change the signal of the time-axis for this the block of every into the signal on a frequency shaft (orthogonal transformation), and it divides into two or more frequency bands. Without blocking the so-called conversion coding method which is a blocking frequency band division method encoded for every band, the audio signal of a time domain, etc. for every unit time amount The band part tally number-ized (sub band coding: SBC) method which is a deblocking frequency band division method divided and encoded to two or more frequency bands can be held. Moreover, the technique and equipment of high efficiency coding which combined the above-mentioned formation of a band part tally number and above-mentioned conversion coding are also considered, and in this case, after the above-mentioned band part tally number-ized method performs band division, orthogonal transformation of the signal for this every band will be carried out to the signal of a frequency domain by the above-mentioned conversion coding method, and it will encode for each [by which orthogonal transformation was carried out] of this band of every.
[0003] As a filter for band division used in the formation of a band part tally number mentioned above here There are filters, such as QMF (Quadrature Mirror filter). For example, the filter of this QMF Reference "digital coding OBU speech Inn subbans" () ["Digital coding of speech in subbands" R.E.Crochiere,] [BellSyst.Tech.] J., Vol.55, and No.8 1976 It is stated. The filter of this QMF is divided into two at bandwidths [band], and in case the band which carried out [above-mentioned] division in the filter concerned is compounded behind, it has been the description that the so-called aliasing does not occur.
[0004] moreover, reference -- "-- poliphase KUADORACHUA fill TAZU - new band part tally number-ized technical" ("Polyphase Quadrature filters-A new subband coding technique", Joseph H.Rothweiler ICASSP 83, BOSTON) -- a poliphase KUADORACHUA filter (Polyphase Quadrature filter) etc. -- etc. -- the filter division technique and equipment of a bandwidth are described. In this poliphase KUADORACHUA filter, in case it divides into two or more bands of bandwidths [signal], it has been the description that it can divide at once.
[0005] Moreover, as orthogonal transformation mentioned above, an input audio signal is blocked with the frame of predetermined unit time amount, and orthogonal transformation which changes a time-axis into a frequency shaft by carrying out discrete Fourier transform (DFT), a discrete cosine transform (DCT), or a MODIFAIDO discrete cosine transform (MDCT) to every block (frame) concerned occurs, for example. in addition, about Above MDCT Reference "time domain aliasing cancellation A basic filter bank design The used subband / conversion coding" () ["Subband/Transform Coding Using Filter Bank Designs Based on Time Domain Aliasing Cancellation,] ["] J. It is stated to P.Princen A.B.Bradley and Univ.of Surrey Royal Melbourne Inst.of Tech.ICASSP 1987.
[0006] Furthermore, as frequency-division width of face in the case of quantizing each frequency component by which frequency band division was carried out, there is band division which took into consideration human being's acoustic-sense property, for example. That is, in the higher region currently generally called the critical band (critical band), bandwidth may divide an audio signal into the band of plurality (for example, 25 bunt) with bandwidth which

becomes large. Moreover, in case the data for every band at this time are encoded, predetermined bit allocation or coding according to accommodative bit allocation the whole band is performed for every band. For example, in case the MDCT multiplier data which MDCT processing was carried out [above-mentioned] and obtained are encoded by the above-mentioned bit allocation, coding will be performed with the accommodative allocation number of bits to the MDCT multiplier data for every band obtained by MDCT processing for every above-mentioned block.
[0007] Two technique and equipment of a degree are known as the above-mentioned bit allocation technique and equipment for it.
[0008] For example, by reference "adaptive transform coding of a sound signal" ("Adaptive Transform Coding of Speech Signals", IEEE Transactions of Accoustics, Speech, and Signal Processing, vol.ASSP-25, No.4, and August 1977), bit allocation is performed based on the magnitude of the signal for every band.
[0009] Moreover, the technique and equipment which obtain the required S/N for every band and perform fixed bit allocation by using auditory masking are described by reference "digital coding about the demand of the perception of a critical band encoder -acoustic-sense system" ("The critical band coder -- digital encoding of theperceptual requirements of the auditory system", and M.A.Kransner MIT and ICASSP 1980), for example.
[0010]
[Problem(s) to be Solved by the Invention] By the way, in the conventional high-efficiency-coding technique and equipment which were mentioned above, there is already a trouble that the system of the quality of loud sound using a higher bit rate cannot be introduced, by the reason the playback machine (decoder) fixed with a certain low bit rate is used, for example.
[0011] Moreover, there is also a trouble that the property of an output signal deteriorates greatly with the property of an input signal, by the reason for the ability not to perform completely optimal bit allocation to the input signal of all properties. That is, when conventional coding equipment is used, ** - this are the cases where coding processing of the input signal which has the change of amplitude information with the case rapid, for example where the quality of an output signal deteriorates greatly depending on the property of the input signal to coding equipment is carried out. In such a case, from conventional sign decryption equipment to an output signal, the noise called the so-called Puri Echo occurs.
[0012] Then, this invention is proposed in view of the above actual condition. Can introduce the system of the quality of loud sound using a higher bit rate which has compatibility also to the playback machine (decoder) fixed with the existing low bit rate, and the input signal of all properties is received. Without being able to perform completely optimal bit allocation and the property of an output signal deteriorating greatly with the property of an input signal It aims at offering high efficiency coding which can obtain the high-efficiency-coding output which Puri Echo does not generate in a decryption equipment side, and a decryption system.
[0013]
[Means for Solving the Problem] High efficiency coding and the decryption system of this invention Two or more coding networks which are proposed in order to attain the object mentioned above, and encode the supplied signal, The high-efficiency-coding equipment possessing the unit or two or more decryption circuits which decrypt the signal by which coding processing was carried out, and the calculus-of-finite-differences appearance circuit which computes difference with the signal by which decryption processing was carried out with the input signal, It comes to have high efficiency decryption equipment possessing two or more decryption circuits which decrypt the signal by which coding processing was carried out, and a synthetic means to compound the signal by which decryption processing was carried out. To the above-mentioned high-efficiency-coding equipment Decryption processing of the coded signal which carried out coding processing of the input signal concerned while carrying out coding processing of the input signal is carried out. Compute the difference of the signal and input signal by which decryption processing was carried out [above-mentioned], and the coding technique which transmits the coded signal which carried out coding processing of the differential signal concerned, and the coded signal which carried out coding processing of the input signal to the above-mentioned high efficiency decryption equipment is applied. It is characterized by applying the decryption technique which compounds on a time-axis and is made into an output signal to the above-mentioned high efficiency decryption equipment, after carrying out decryption processing of the coded signal by which transmission was carried out [above-mentioned].
[0014] Here, the circuit where using the same circuit or a different circuit and/or two or more decryption circuits in the above-mentioned high-efficiency-coding equipment, or two or more decryption circuits in the above-mentioned high efficiency decryption equipment are also the same, or a different circuit can be used for two or more coding networks in the above-mentioned high-efficiency-coding equipment. Moreover, the above-mentioned high efficiency decryption equipment can perform decryption processing only using some coded signals of the differential signal of

the coded signal of the above-mentioned input signal sent from the above-mentioned high-efficiency-coding equipment and plurality, or an unit, or the coded signal of the above-mentioned input signal. Furthermore, it performs that the above-mentioned high-efficiency-coding equipment's outputting the coded signal of a singular input signal and the coded signal of two or more above-mentioned differential signals and/or the above-mentioned high efficiency decryption equipment carry out decryption processing, and compound the coded signal of the coded signal of the input signal of the unit of high-efficiency-coding equipment, and two or more above-mentioned differential signals.

[0015] Moreover, in high efficiency coding and the decryption system of this invention, the coding network in the above-mentioned high-efficiency-coding equipment shall perform nonlinear quantization, and the decryption circuit in the above-mentioned high-efficiency-coding equipment and high efficiency decryption equipment shall use nonlinear reverse quantization only for performing nonlinear reverse quantization or coding of the differential signal in the above-mentioned high-efficiency-coding equipment only at a decryption of the coded signal of the differential signal in high efficiency decryption equipment using nonlinear quantization. Furthermore, reversible sign decryption processing is performed in the coding network of the differential signal in the above-mentioned high-efficiency-coding equipment, and the decryption circuit of the coded signal of the differential signal in high efficiency decryption equipment. Moreover, the coding network of the above-mentioned high-efficiency-coding equipment computes the degree of a masking effect, the die length of each processing block is determined, and the coding network of the above-mentioned differential signal performs bit assignment which depended more mostly to the acoustic-sense permissible noise spectrum. Furthermore, the coding network of the above-mentioned high-efficiency-coding equipment divides the signal on a frequency shaft to two or more bands while changing the signal on a time-axis into the signal on a frequency shaft by orthogonal transformation, and the decryption circuit of the above-mentioned high efficiency decryption equipment changes the signal of two or more bands on a frequency shaft into the signal on a time-axis by reverse orthogonal transformation. A reverse modification discrete cosine transform (IMDCT) is used as reverse orthogonal transformation, using a modification discrete cosine transform (MDCT) as the above-mentioned orthogonal transformation at this time. Furthermore, the coding network of the above-mentioned high-efficiency-coding equipment divides the signal of the single band on a time-axis into the signal of two or more bands with a mulberry DORACHA mirror filter (QMF), and the decryption circuit of the above-mentioned high efficiency decryption equipment compounds the signal divided into two or more bands on a time-axis with the in berth mulberry DORACHA mirror filter (IQMF) to the signal of a single band. Moreover, the above-mentioned high-efficiency-coding equipment and high efficiency decryption equipment can process the signal for two or more channels.

[0016] Next, high efficiency coding and the decryption system of this invention have the transmission medium which records or transmits the encoded signal, and the above-mentioned high-efficiency-coding equipment at this time separates two or more coded signals to output in one sink block, and is recorded or transmitted to the above-mentioned transmission medium.

[0017]

[Function] If according to this invention the difference of an input signal and the signal which carried out decryption processing after carrying out coding processing of this input signal is a noise component generated by coding and a decryption of an input signal, therefore the coded signal of this input signal and the coded signal of a differential signal are decrypted with high efficiency decryption equipment and it compounds on a time-axis, it will become possible to make property degradation of the output signal depending on the property of an input signal mitigate.

[0018] moreover, even when the decryption circuit fixed with the low bit rate which already exists in high efficiency decryption equipment, for example is used according to this invention By preparing two or more decryption circuits in high efficiency decryption equipment, performing decryption processing in each decryption circuit, and compounding each output signal (coded signal) of the high-efficiency-coding equipment of this invention on a time-axis It becomes possible to decrease the quality difference of the input signal of high-efficiency-coding equipment, and the output signal of high efficiency decryption equipment, using the existing decryption circuit.

[0019]

[Example] Hereafter, with reference to a drawing, the example of the high-efficiency-coding equipment (encoder) which constitutes high efficiency coding and the decryption system of this invention, and high efficiency decryption equipment (decoder) is explained.

[0020] The configuration of one example of high efficiency coding of this invention and a decryption system is shown in drawing 1 . (a) of this drawing 1 shows the configuration of the high-efficiency-coding equipment of this invention, and (b) of drawing 1 shows the configuration of the high efficiency decryption equipment of this invention.

[0021] Namely, high efficiency coding and the decryption system of this invention example Two or more coding networks 51 and 54 which encode the supplied signal as shown in (a) of drawing 1 , the difference which computes the difference of the unit or two or more decryption circuits 52 which decrypt the signal by which coding processing was carried out, and the signal by which decryption processing was carried out with the input signal, as it is indicated in (b) of drawing 1 as the high-efficiency-coding equipment possessing the calculation circuit 53 It comes to have high efficiency decryption equipment possessing two or more decryption circuits 59 and 60 which decrypt the signal by which coding processing was carried out, and the adder circuit 61 as a synthetic means to compound the signal by which decryption processing was carried out. Decryption processing of the coded signal which carried out coding processing of the input signal concerned here at the above-mentioned high-efficiency-coding equipment while carrying out coding processing of the input signal is carried out. Compute the difference of the signal and input signal by which decryption processing was carried out [above-mentioned], and the coding technique which transmits the coded signal which carried out coding processing of the differential signal concerned, and the coded signal which carried out coding processing of the input signal to the above-mentioned high efficiency decryption equipment is applied. After carrying out decryption processing of the coded signal by which transmission was carried out [above-mentioned], he is trying to apply the decryption technique which compounds on a time-axis and is made into an output signal to the above-mentioned high efficiency decryption equipment.
[0022] First, in (a) of drawing 1 , the 0-22kHz audio PCM signal for two or more channels is supplied to the high-efficiency-coding equipment input terminal 50 of this invention example. This input signal is sent to a coding network 51 and the calculus-of-finite-differences appearance circuit 53.
[0023] A coding network 51 has a configuration equivalent to conventional coding equipment, and performs high efficiency coding for input digital signals, such as an audio signal, using each technique of the formation (SBC) of a band part tally number, adaptive transform coding (ATC), and adaptation bit allocation (APC-AB). The output signal of a coding network 51 is sent to the high-efficiency-coding equipment output terminal 55 and the decryption circuit 52.
[0024] The decryption circuit 52 has a configuration equivalent to conventional decryption equipment, decrypts the signal encoded by the above-mentioned coding network 51, and returns it to an audio PCM signal etc. In addition, this audio PCM signal includes the noise generated by the coding network 51 and the decryption circuit 52. The signal decrypted here is sent to the calculus-of-finite-differences appearance circuit 53.
[0025] The calculus-of-finite-differences appearance circuit 53 computes reception and its differential signal for two kinds of audio PCM signals different, respectively from an input terminal 50 and the decryption circuit 52. The differential signal computed here means the noise component generated by the coding network 51 and the decryption circuit 52. This differential signal is sent to a coding network 54.
[0026] Like a coding network 51, a coding network 54 has a configuration equivalent to conventional coding equipment, and performs high efficiency coding to the differential signal computed by the calculus-of-finite-differences appearance circuit 53. The output signal of a coding network 54 is sent to the high-efficiency-coding equipment output terminal 56.
[0027] In addition, by making this coding network 54 and the coding network 51 of the preceding paragraph into the same circuit, system magnitude of high-efficiency-coding equipment can be made small, and it becomes possible to stop the consumed electric power of cost or hardware low. Moreover, when the high-efficiency-coding circuit is already existing, it becomes possible to divert the coding network as it is.
[0028] Next, in (b) of drawing 1 , the high-efficiency-coding signal is supplied to the high efficiency decryption equipment input terminal 57 of this invention from the high-efficiency-coding equipment output terminal 55. This input signal is sent to the decryption circuit 59.
[0029] The decryption circuit 59 has a configuration equivalent to conventional decryption equipment, decrypts the coded signal sent from the high-efficiency-coding equipment output terminal 55, and returns it to an audio PCM signal etc. In addition, this decryption circuit 59 has a configuration equivalent to the decryption circuit 52 in (a) of drawing 1 . That is, the output signal of the decryption circuit 52 and the output signal of the decryption circuit 59 are equivalent. The signal decrypted here is sent to an adder circuit 61.
[0030] Moreover, the high-efficiency-coding signal is supplied to the high efficiency decryption equipment input terminal 58 of this invention from the high-efficiency-coding equipment output terminal 56. This input signal is sent to the decryption circuit 60.
[0031] Like the decryption circuit 59, the decryption circuit 60 has a configuration equivalent to conventional decryption equipment, decrypts the coded signal sent from the high-efficiency-coding equipment output terminal 56, and returns it to an audio PCM signal etc. The signal decrypted here is sent to an adder circuit 61.

[0032] For example, when the coding networks 51 and 54 in drawing 1 are circuits of the completely same configuration, also as for the decryption circuits 52, 59, and 60 in drawing 1 , it is most effective to use the circuit of the same configuration. This can make small system magnitude of the high efficiency decryption equipment of this invention, and becomes possible [stopping the consumed electric power of cost or hardware low].

[0033] Furthermore, when the high-efficiency-coding circuit which is existing to the coding networks 51 and 54 in drawing 1 is used as mentioned above, it becomes possible to divert the high efficiency decryption circuit which is existing also to the decryption circuits 52, 59, and 60 in drawing 1 . Thereby, in order to consume the coding network of the same specification, and a decryption circuit to a large quantity, the unit price of a circuit falls and it becomes possible to lower the production cost of the high efficiency decryption equipment with which with equipment, and this invention is existing. [equipment / high efficiency coding / and] [decryption]

[0034] Moreover, in the high-efficiency-coding equipment of this invention, and decryption equipment, using existing coding and a decryption circuit means having the compatibility over existing high-efficiency-coding equipment and decryption equipment. For example, a sign and decrypting can be realized, securing tone quality equivalent to the present high efficiency sign decryption equipment by not receiving the coded signal outputted from the high-efficiency-coding equipment output terminal 56 in drawing 1 , but receiving only the coded signal outputted from the high-efficiency-coding equipment output terminal 55, in case the signal encoded with the high-efficiency-coding equipment of this invention is decrypted with existing high efficiency decryption equipment. Moreover, a sign and decrypting become realizable, securing tone quality equivalent to the present high efficiency sign decryption equipment by inputting a coded signal into one of the high efficiency decryption equipment input terminals 57 and 58 in drawing 1 , in case the signal encoded with existing high-efficiency-coding equipment is decrypted with the high efficiency decryption equipment of this invention. Moreover, for example, the same coded signal as the high efficiency decryption equipment input terminals 57 and 58 in drawing 1 can be inputted, and it can realize also by adding the function which chooses whether it adds to an adder circuit 61.

[0035] An adder circuit 61 adds two kinds sent from the decryption circuits 59 and 60 of decrypted signals. By performing addition in this adder circuit 61, generating of the quantizing noise generated in processing of a coding network 51 and the decryption circuit 59 is controlled. namely, the signal and quality by which the signal outputted from the output terminal 62 of high efficiency decryption equipment is supplied to the input terminal 50 of high-efficiency-coding equipment -- abbreviation -- it will be obtained as the same thing. Especially, in processing of a coding network 51 and the decryption circuit 59, it is effective to a specific input signal which emits a loud quantizing noise. The signal after addition processing is sent to the high efficiency decryption equipment output terminal 62.

[0036] From the high efficiency decryption equipment output terminal 62, the audio PCM signal decrypted by the high efficiency decryption equipment of this invention is outputted.

[0037] In this example, high efficiency coding is performed for input digital signals, such as an audio PCM signal, using each technique of the formation (SBC) of a band part tally number, adaptive transform coding (ATC), and adaptation bit allocation (APC-AB) in the coding network 51 and coding network 54 in drawing 1 . This technique is explained referring to drawing 2 .

[0038] With the concrete high-efficiency-coding equipment of this example shown in drawing 2 , while a filter etc. divides an input digital signal into two or more frequency bands, orthogonal transformation was performed for every frequency band, and to every [in consideration of the acoustic-sense property of human being who mentions the spectrum data of the acquired frequency shaft later] so-called critical band width of face (critical band), bit allocation was carried out accommodative and it has encoded. At this time, the band which divided critical band width of face further is used in a high region. Of course, the frequency-division width of face of not blocking according to a filter etc. is good also as division-into-equal-parts ****.

[0039] Furthermore, in this invention example, while changing a block size (block length) accommodative according to an input signal before orthogonal transformation, the small block which subdivided further critical band width of face (critical band) is performing floating processing in the critical band unit or the high region. In addition, this critical band is the frequency band divided in consideration of human being's acoustic-sense property, and is a band which that noise in case the mask of the pure sound concerned is carried out by the narrow-band band noise of the same strength near the frequency of a certain pure sound has. The perimeter wave number band whose bandwidth this critical band is large like the high region, for example, is 0-20kHz is divided into the critical band of 25.

[0040] That is, in drawing 2 , the 0-22kHz audio PCM signal is supplied to the input terminal 10. This input signal is divided into a 0-11kHz band and 11k-22kHz band by the band division filters 11, such as the so-called QMF mentioned above, and, similarly the signal of a 0-11kHz band is divided into a 0-5.5kHz band and a 5.5k-11kHz band by the band division filters 12, such as the so-called QMF.

[0041] The signal of 11k-22kHz band from the above-mentioned band division filter 11 is sent to the MDCT (Modified Discrete Cosine Transform) circuit 13 which is an example of a rectangular conversion circuit, the signal of the 5.5k-11kHz band from the above-mentioned band division filter 12 is sent to the MDCT circuit 14, and MDCT processing of the signal of the 0-5.5kHz band from the above-mentioned band division filter 12 is carried out by being sent to the MDCT circuit 15, respectively. In addition, in each MDCT circuits 13, 14, and 15, MDCT processing is made based on the block size determined by the block decision circuits 19, 20, and 21 prepared for every band.

[0042] Here, the example of the block size in each MDCT circuits 13, 14, and 15 determined by the above-mentioned block decision circuits 19, 20, and 21 is shown in A and B of drawing 3 . In addition, the case (orthogonal transformation block size in short mode) where an orthogonal transformation block size is short is shown for the case (orthogonal transformation block size in long mode) where an orthogonal transformation block size is long in A of drawing 3 at B of drawing 3 .

[0043] In the example of this drawing 3 , three filter outputs have two orthogonal transformation block sizes, respectively. That is, to the signal of the 0-5.5kHz band by the side of low-pass, and the signal of the 5.5k-11kHz band of a mid-range, in the case of the long block length (A of drawing 2), the measurement size in 1 block is made into 128 samples, and when a short block is chosen (B of drawing 3), the measurement size in 1 block is considered as the block for every 32 samples. On the other hand, to the signal of 11k-22kHz band by the side of a high region, in the case of the long block length (A of drawing 3), the measurement size in 1 block is made into 256 samples, and when a short block is chosen (B of drawing 3), the measurement size in 1 block is considered as the block for every 32 samples. thus -- being the same in the measurement size of an orthogonal transformation block of each band, when a short block is chosen -- carrying out -- like a high region -- time resolution -- raising -- in addition -- and the class of window used for blocking is reduced.

[0044] In addition, the information which shows the block size determined in the above-mentioned block decision circuits 19, 20, and 21 is outputted from output terminals 23, 25, and 27 while it is sent to the below-mentioned adaptation bit allocation code-ized circuits 16, 17, and 18.

[0045] Again, the spectrum data or MDCT multiplier data of a frequency domain which MDCT processing was carried out in drawing 2 in each MDCT circuits 13, 14, and 15, and was obtained is gathered for every band which divided the critical band further, and is sent to the adaptation bit allocation code-ized circuits 16, 17, and 18 in the so-called critical band (critical band) or the high region.

[0046] In the adaptation bit allocation code-ized circuits 16, 17, and 18, it is made to carry out re-quantization (for it to normalize and quantize) of each spectrum data (or MDCT multiplier data) according to the number of bits which divided the critical band further and which was assigned for every band in the information and critical band (critical band), or high region of the above-mentioned block size.

[0047] The data encoded by each [these] adaptation bit allocation code-ized circuits 16, 17, and 18 are taken out through output terminals 22, 24, and 26. Moreover, in the adaptation bit allocation code-ized circuits 16, 17, and 18 concerned, the scale factor which shows whether the normalization about the magnitude of what kind of signal was made, and the bit length information which shows by what kind of bit length quantization was carried out are also searched for, and these are also simultaneously outputted from output terminals 22, 24, and 26.

[0048] Moreover, from the output of each MDCT circuits 13, 14, and 15 in drawing 2 , the energy for every band which divided the critical band further is searched for in the above-mentioned critical band (critical band) or a high region by calculating the square root of the root mean square of each amplitude value within the band concerned, for example etc. Of course, you may make it use for future bit allocation of the above-mentioned scale factor itself. In this case, since the operation of new energy count becomes unnecessary, it becomes economization of hard magnitude. Moreover, it is also possible to use the peak value of amplitude value, the average, etc. instead of the energy for every band.

[0049] Next, in the adaptation bit quota circuits 16, 17, and 18, the concrete technique of the above-mentioned bit allocation is explained.

[0050] If drawing 4 explains actuation of the adaptation bit allocation circuit in this case, the magnitude of a MDCT multiplier will be called for for every block, and that MDCT multiplier will be supplied to an input terminal 801. The MDCT multiplier supplied to the input terminal 801 concerned is given to the energy calculation circuit 803 for every band. In the energy calculation circuit 803 for every band, the signal energy about each band which re-divided the critical band further in the critical band or the high region is computed. The energy about each band computed in the energy calculation circuit 803 for every band is supplied to the energy dependence bit allocation circuit 804.

[0051] In the energy dependence bit allocation circuit 804, it comes to perform bit allocation which makes white

quantizing noise using a certain rate of the 128Kbps in the usable total bit by the usable total bit generating circuit 802, and this example. The rate that this amount of bits occupies to the above-mentioned 128Kbps increases, so that unevenness of the spectrum of an input signal is so large that the toe nullity of an input signal is high at this time. In addition, in order to detect unevenness of the spectrum of an input signal, the sum of the absolute value of the difference of the block floating multiplier of an adjoining block is used as an index so that it may mention later. And after that, bit allocation proportional to the pair numeric value of the energy of each band is performed about the calculated usable amount of bits so that it may mention later.

[0052] The bit allocation calculation circuit 805 depending on acoustic-sense allowance noise level calculates the amount of allowance noises for every critical band in consideration of the so-called masking effect etc. based on the spectrum data first divided for every above-mentioned critical band, and a part for the bit which lengthened the energy dependence bit from the above-mentioned usable total bit so that an acoustic-sense permissible noise spectrum might next be given is distributed. Thus, the called-for energy dependence bit and the bit depending on acoustic-sense allowance noise level are added, and each spectrum data (or MDCT multiplier data) is re-quantized by the adaptation bit allocation code-ized circuits 16, 17, and 18 of drawing 2 according to the number of bits assigned to the band which divided the critical band into two or more bands further in every critical band and the high region. Thus, the encoded data are taken out through the output terminals 22, 24, and 26 of drawing 2 .

[0053] If the acoustic-sense permissible noise spectrum calculation circuit in the bit allocation circuit 805 of the above-mentioned acoustic-sense permissible noise spectrum dependence is explained in more detail, the MDCT multiplier obtained in the MDCT circuits 13, 14, and 15 will be given to the permissible noise spectrum calculation circuit in the bit allocation circuit 805 concerned.

[0054] Drawing 5 is the block circuit diagram showing the outline configuration of one example in which the above-mentioned permissible noise spectrum calculation circuit was explained collectively. In this drawing 5 , the spectrum data of the frequency domain from the MDCT circuits 13, 14, and 15 are supplied to the input terminal 521.

[0055] The input data of this frequency domain is sent to the energy calculation circuit 522 for every band, and when the energy of every above-mentioned critical band (critical band) calculates total of each amplitude value square within the band concerned, for example, it is called for. The peak value of amplitude value, the average, etc. may be used instead of the energy for every band of this. Generally as an output from this energy calculation circuit 522, the spectrum of the total value of for example, each band is called the bark spectrum. Drawing 6 shows the bark spectrum SB for such every critical band. However, in this drawing 6 , in order to simplify a graphic display, 12 bands (B1 - B12) are expressing the number of bands of the above-mentioned critical band.

[0056] Reefing (convolution) processing which hangs and adds a predetermined weighting function to this bark spectrum SB here in order to take into consideration the effect in the so-called masking of the above-mentioned bark spectrum SB is performed. For this reason, each value of the output SB of the energy calculation circuit 522 for every above-mentioned band, i.e., this bark spectrum, is sent to the reefing filter circuit 523. This reefing filter circuit 523 consists of two or more delay elements which carry out sequential delay of the input data, two or more multipliers (for example, 25 multipliers corresponding to each band) which carry out the multiplication of the filter coefficient (weighting function) to an output from these delay elements, and a total adder which takes total of each multiplier output.

[0057] In addition, the above-mentioned masking says the phenomenon which the mask of other signals is carried out by a certain signal, and stops being able to hear with the property on human being's acoustic sense, and there are a time-axis masking effect by the audio signal of a time domain and this time-of-day masking effect by the signal of a frequency domain as this masking effect. According to these masking effects, even if a noise is in the part masked, this noise can be heard. For this reason, let the noise within the limits of [which is masked] this be a permissible noise in a actual audio signal.

[0058] Moreover, if one example of the multiplication multiplier (filter coefficient) of each multiplier of the above-mentioned reefing filter circuit 523 is shown, when setting the multiplier of the multiplier M corresponding to the band of arbitration to 1, A multiplier 0.15 with a multiplier M-2 with a multiplier M-1 a multiplier 0.0019 In a multiplier 0.0000086, reefing processing of the above-mentioned bark spectrum SB is performed by the multiplier M-3 with a multiplier M+1 by carrying out [a multiplier 0.4] the multiplication of the multiplier 0.007 for a multiplier 0.06 to the output of each delay element with a multiplier M+3 with a multiplier M+2. However, M is the integer of the arbitration of 1-25.

[0059] Next, the output of the above-mentioned reefing filter circuit 523 is sent to the subtraction machine 524. This subtraction machine 524 asks for the level alpha corresponding to the noise level in which the allowance in the collapsed field mentioned later is possible the account of a top. In addition, the level alpha corresponding to the noise

level (allowance noise level) in which the allowance concerned is possible is the level which turns into an allowance noise level for every band of a critical band by performing reverse convolution processing, as mentioned later.

[0060] Here, the admissible function (function expressing masking level) of the ** sake which asks for the above-mentioned level alpha is supplied to the above-mentioned subtraction machine 524. The above-mentioned level alpha is controlled by making this admissible function fluctuate. The admissible function concerned is supplied from the function generating circuit (n-ai) 525 which is explained below.

[0061] That is, if the number given sequentially from low-pass [of the band of a critical band] is set to i, it can ask for the level alpha corresponding to an allowance noise level by the following formula.

[0062] Alpha=S - (n-ai)

In this formula, n and a are the reinforcement of the bark spectrum with which reefing processing of a> 0 and the S was carried out by the constant, and the inside (n-ai) of a formula serves as an admissible function. n= 38 and a=-0.5 can be used as an example.

[0063] Thus, the above-mentioned level alpha is called for and this data is transmitted to a divider 526. It is for carrying out the reverse convolution of the above-mentioned level alpha in the field by which the reefing was carried out [above-mentioned] in the divider 526 concerned. Therefore, Massu Kings RESSHORUDO comes to be obtained from the above-mentioned level alpha by performing this reverse convolution processing. That is, this Massu Kings RESSHORUDO serves as an allowance noise spectrum. In addition, although the above-mentioned reverse convolution processing needs a complicated operation, it is performing the reverse convolution using the simplified divider 526 in this example.

[0064] Next, above-mentioned Massu Kings RESSHORUDO is transmitted to a subtractor 528 through the synthetic circuit 527. Here, the bark spectrum SB outputted namely, mentioned above from the energy detector 522 for every above-mentioned band is supplied to the subtractor 528 concerned through the delay circuit 529. Therefore, as shown in drawing 7 by the subtraction operation of above-mentioned Massu Kings RESSHORUDO and the bark spectrum SB being performed with this subtractor 528, below the level that shows the above-mentioned bark spectrum SB on the level of Massu Kings RESSHORUDO MS concerned will be masked. In addition, the above-mentioned delay circuit 529 is formed in order to delay the bark spectrum SB from the energy detector 522 in consideration of the amount of delay in each circuit before the above-mentioned synthetic circuit 527.

[0065] The output from the subtractor 528 concerned is sent to ROM with which it was taken out through the output terminal 531, for example, allocation number-of-bits information was beforehand remembered to be through the permissible noise amendment circuit 530 (not shown). This ROM etc. outputs the allocation number-of-bits information for every band according to the output (level of the difference of the energy of each above-mentioned band, and the output of the above-mentioned noise level setting-out means) obtained from the above-mentioned subtractor circuit 528 through the permissible noise amendment circuit 530.

[0066] Thus, an energy dependence bit and the bit depending on acoustic-sense allowance noise level are that are added and the allocation number-of-bits information is sent to the above-mentioned adaptation bit allocation code-ized circuits 16, 17, and 18, and each spectrum data of the frequency domain from the MDCT circuits 13, 14, and 15 is quantized here with the number of bits assigned for every band.

[0067] That is, if it summarizes, in the above-mentioned adaptation bit allocation code-ized circuits 16, 17, and 18, the spectrum data for every above-mentioned band will be quantized with the number of bits distributed according to the level of the difference of the energy of a band or peak value which divided the critical band concerned into two or more bands further in every band band (every critical band) and high region of the above-mentioned critical band, and the output of the above-mentioned noise level setting-out means.

[0068] By the way, in the case of composition, the data in which the so-called minimum audible curve RC which is human being's acoustic-sense property as shown in drawing 8 in the synthetic circuit 527 mentioned above supplied from the minimum audible curve generating circuit 532 is shown, and above-mentioned Massu Kings RESSHORUDO MS are compoundable. In this minimum audible curve, if noise absolute level becomes below this minimum audible curve, this noise can be heard. although it becomes what is different by the difference in the playback volume at the time of playback even if this minimum audible curve has the same coding for example, -- a realistic digital system -- the sound for example, to a 16-bit dynamic range -- since there is no difference in the easy method of entering so much, supposing the quantizing noise of the frequency band near 4kHz a lug is the easiest to hear cannot be heard, for example, in other frequency bands, it will be thought that the quantizing noise below the level of this minimum audible curve cannot be heard. Therefore, it assumes that usage depending on which the noise near 4kHz of the dynamic range which a system has in this way cannot be heard is carried out, and if an allowance noise level is obtained by compounding both this minimum audible curve RC and Massu Kings RESSHORUDO MS,

the allowance noise level in this case can be carried out to to the part shown with the slash in drawing 8 . In addition, in this example, the level of 4kHz of the above-mentioned minimum audible curve is doubled with the minimum level of 20 bits. Moreover, this drawing 8 shows the signal spectrum SS simultaneously.

[0069] Moreover, in the above-mentioned permissible noise amendment circuit 530, the allowance [in / based on the information on an equal loudness curve for example / the output from the above-mentioned subtractor 528] noise level sent from the amendment information output circuit 533 is amended. It connected with the curve in quest of the sound pressure of the sound in each frequency which it is here, and an equal loudness curve is a characteristic curve about human being's acoustic-sense property, for example, the same magnitude as the pure sound of 1kHz hears, and is also called the equal loudness contour of loudness rating. moreover, the minimum audible curve RC which showed these loudness level contours to drawing 8 and abbreviation -- the same curve is drawn. In these loudness level contours, for example near 4kHz, the magnitude as 1kHz also with the same bottom of 8-10dB hears sound pressure from the place of 1kHz, and near 50Hz, unless it is higher than the sound pressure in 1kHz about 15dB conversely, the same magnitude does not hear at it. For this reason, the noise (allowance noise level) beyond the level of the above-mentioned minimum audible curve is understood are good to have the frequency characteristics given in the curve according to these loudness level contours. Since it is such, it turns out that amending the above-mentioned allowance noise level in consideration of the above-mentioned loudness level contours conforms to human being's acoustic-sense property.

[0070] The spectrum configuration depending on the acoustic-sense allowance noise level described above is built with the bit allocation using a certain rate of the usable total bit 128Kbps. This rate decreases, so that the toe nullity of an input signal becomes high.

[0071] Next, the amount division technique of bits between the two bit allocation technique is explained.

[0072] It returns to drawing 4 , and the signal from the input terminal 801 with which a MDCT circuit output is supplied is also given to the smoothness calculation circuit 808 of a spectrum, and the smoothness of a spectrum is computed here. In this example, the value which broke the sum of the absolute value of the difference between the neighbors of the absolute value of a signal spectrum by the sum of the absolute value of a signal spectrum is computed as smoothness of the above-mentioned spectrum.

[0073] The output of the smoothness calculation circuit 808 of the above-mentioned spectrum is given to the rate decision circuit 809 of bit division, and the rate of bit division between bit allocation of energy dependence and the bit allocation by the acoustic-sense permissible noise spectrum is determined here. It thinks that the rate of bit division does not have the smoothness of a spectrum, so that the output value of the smoothness calculation circuit 808 of a spectrum is large, and bit allocation with emphasis on the bit allocation by the acoustic-sense permissible noise spectrum is performed rather than bit allocation of energy dependence. The rate decision circuit 809 of bit division sends a control output to the multipliers 811 and 812 which control the magnitude of bit allocation of energy dependence, and the bit allocation by the acoustic-sense permissible noise spectrum, respectively. Here, a spectrum is temporarily smooth, and when the output of the rate decision circuit 809 of bit division to a multiplier 811 takes the value of 0.8 so that weight may be set to bit allocation of energy dependence, the output of the rate decision circuit 809 of bit division to a multiplier 812 is set to 1-0.8=0.2. The output of these two multipliers is added by the adder 806, serves as final bit allocation information, and is outputted from an output terminal 807.

[0074] The situation of the bit allocation at this time is shown in drawing 9 and drawing 10 . Moreover, the situation of the quantizing noise corresponding to this is shown in drawing 11 and drawing 12 . Drawing 9 shows the case where the spectrum of a signal is comparatively flat, and drawing 10 shows the case where a signal spectrum shows a high toe nullity. Moreover, the amount of bits for signal level dependence is shown by the inside QS of drawing of drawing 9 and drawing 10 , and the inside QN of drawing shows the amount of bits for bit allocation of acoustic-sense allowance noise level dependence. Signal level is shown by the inside L of drawing of drawing 11 and drawing 12 , and the inside NS of drawing shows a part for noise lowering according [the inside NN of drawing] a part for the noise lowering by part for signal level dependence to a part for bit allocation of acoustic-sense allowance noise level dependence.

[0075] First, the bit allocation for which the spectrum of a signal depended on acoustic-sense allowance noise level in drawing 11 which shows the case of being comparatively flat is useful in order to cross to all bands and to take large signal to noise ratio. However, comparatively little bit allocation is used in low-pass and a high region. This is because the sensibility to the noise of this band is small in acoustic sense. Although there are few parts of bit allocation depending on a signal energy level as an amount, it is preponderantly distributed to the frequency domain where inside low-pass signal level is high in this case so that a White noise spectrum may be produced.

[0076] On the other hand, as shown in drawing 12 , when a signal spectrum shows a high toe nullity, the amount of

bit allocation depending on a signal energy level increases, and lowering of a quantizing noise is used in order to reduce the noise of a narrow band extremely. Concentration of the bit allocation depending on acoustic-sense allowance noise level is not tighter than this.

[0077] As shown in drawing 12 , improvement in the property in an isolated spectrum input signal is attained by the sum of these both bit allocation.

[0078] Drawing 13 shows the decryption equipment of the fundamental this invention example for decrypting the encoded signal again in the decryption circuits 52, 59, and 60 in drawing 1 .

[0079] In this drawing 13 , the MDCT multiplier by which each band was quantized is given to the decryption equipment input terminals 122, 124, and 126, and the used block-size information is given to input terminals 123, 125, and 127. In the decryption circuits 116, 117, and 118, bit allocation is canceled using adaptation bit allocation information.

[0080] Next, the signal of a frequency domain is changed into the signal of a time domain in the IMDCT circuits 113, 114, and 115. The time domain signal of these partial bands is decrypted by the whole region signal, and is sent to an output terminal 110 by the IQMF circuits 112 and 111.

[0081] Next, the signal encoded by the high-efficiency-coding equipment of this invention example which mentioned above, the transmission medium, i.e., the media, of this invention example, is recorded or transmitted. That is, record is also included as transmission said here. What the above-mentioned coded signal was recorded on the record medium of the shape of a disk, such as an optical disk, a magneto-optic disk, and a magnetic disk, as media for record, the thing by which the above-mentioned coded signal was recorded on tape-like record media, such as a magnetic tape, or the semiconductor memory the coded signal was remembered to be, an IC card, etc. can be mentioned. Moreover, as media for transmission which does not include record, an electric wire or an optical cable, an electric wave, etc. can be mentioned.

[0082] In addition, about arrangement of the data in the media of this invention example, it becomes an array as shown, for example in (A) in drawing 14 , and (B). namely, one sink block -- sink information IS Sub information ISB outputted from the sub information ISA (the scale factor, WORD length) and the Maine information IMA which were outputted from the high-efficiency-coding equipment output terminal 55 in drawing 1 , and the high-efficiency-coding equipment output terminal 56 Maine information IMB from -- it shall become

[0083] In this case, it is effective to dissociate, to record or transmit the signal sent from each high-efficiency-coding equipment output terminals 55 and 56 into one sink block, and to decode and reproduce after that at the point that the bit string part which should be removed can be removed collectively, when lowering a bit rate and reproducing using existing high efficiency decryption equipment.

[0084] moreover -- for example, in case the data of a certain capacity recorded on media are copied on another media, in order to save the capacity of media and to realize more nearly prolonged record, it is effective at the point that the bit string part which should be removed can be removed collectively to record what removed the coded signal outputted from the high-efficiency-coding equipment output unit 56.

[0085] For example, especially the above bit arrays used the magneto-optic disk and the optical disk, they are applicable to the so-called mini disc (Mini Disc), magnetic tape media, a communication medium, etc.

[0086] In addition, it is also possible to make it a configuration which this invention is not limited only to this example and is different in two coding networks 51 and 54 in drawing 1 . Thereby, although the decryption circuit 52 and the decryption circuit 59 in drawing 1 are the same configuration, the decryption circuit 60 serves as a different configuration. That is, a coding network 51 has the relation to which the decryption circuits 52 and 59 and a coding network 54 corresponded with the decryption circuit 60, respectively. For example, it is possible to use for another coding network the band part tally number-ized method which is a deblocking frequency band division method at one of coding networks using the conversion coding method which is a blocking frequency band division method. In addition, since many so-called white-noise components are included (for example, since it is easy to carry out bit allocation of the direction which used for the coding network 54 the band part tally number-ized method currently divided into the same bandwidth uniformly to a perimeter wave number band), the input signal of a coding network 54 acts on validity more.

[0087] Moreover, in the high-efficiency-coding equipment of (a) of drawing 1 , it is also possible to have the structure which repeats a series of processings performed by coding networks 51 and 54, the decryption circuit 52, and the calculus-of-finite-differences appearance circuit 53 two or more times. That is, it is also possible to take two or more steps of layered structures of decrypting again the signal encoded by the coding network 54, computing difference with the input signal from the high-efficiency-coding equipment input terminal 50, and encoding it further. In addition, in connection with it, high efficiency decryption equipment serves as a configuration which increases a

decryption circuit.

[0088] Moreover, since the input signal over the coding network 54 in drawing 1 is a differential signal of the signal supplied from the high-efficiency-coding equipment input terminal 50, and the signal by which sign decryption processing was carried out in the coding network 51 and the decryption circuit 52, it contains many so-called components of white noise. Therefore, the quantization distortion in re-quantization can be suppressed by using a nonlinear quantizer to the coding network 54 and the decryption circuit 60 in drawing 1 .

[0089] Further for example, when a reversible sign decryption method is used to the coding network 54 in drawing 1 , and the decryption circuit 60, the high-efficiency-coding equipment input signal of this invention and a high efficiency decryption equipment output signal turn into the same signal, and the whole system serves as a reversible sign decryption method.

[0090] Moreover, when it has the same configuration as the coding method explained, for example in this example, it may act on validity more by changing the various setting-out parameters in a coding network. For example, when using the high efficiency decryption equipment of a configuration when a bit rate higher than a coding network 54 is set up to the coding network 51 in drawing 1 by changing the parameter of the usable total number-of-bits generating circuit 802 in drawing 4 , so that it may decrypt only using the coded signal inputted from the high efficiency decryption equipment input terminal 57 in drawing 1 , it acts effectively.

[0091] Moreover, since the input signal over the coding network 54 in drawing 1 is a differential signal of the signal supplied from the high-efficiency-coding equipment input terminal 50, and the signal by which sign decryption processing was carried out in the coding network 51 and the decryption circuit 52, it contains many so-called components of white noise. Therefore, in a coding network 54, it becomes possible, for example by changing a parameter so that more bits by the bit allocation 805 of an acoustic-sense permissible noise spectrum may be distributed in the rate decision circuit 809 of bit division in drawing 4 to perform bit allocation which was adapted for the input signal to a coding network 54.

[0092] this invention example can consider the above various deformation.

[0093]

[Effect of the Invention] In high efficiency coding and the decryption system of this invention, the following effectiveness can be acquired so that clearly also from the above explanation.

[0094] That is, even when the coding equipment and decryption equipment using a low bit rate are already used for the 1st, in case it is going to introduce coding of the quality of loud sound using a higher bit rate, and a decryption system, the system which has compatibility with existing coding equipment and decryption equipment can be offered.

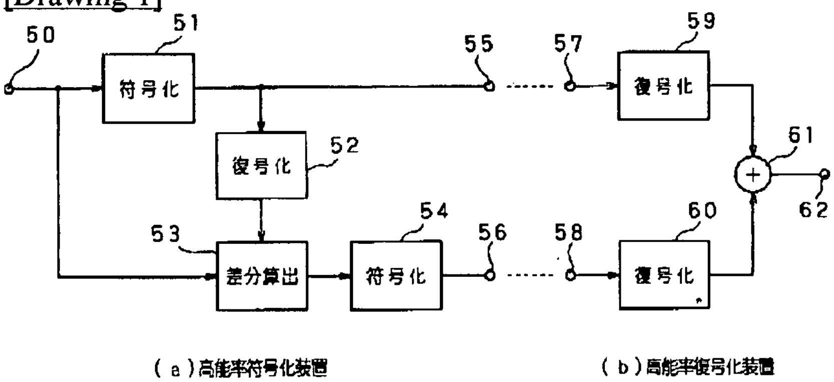
[0095] high [2nd] -- since tone quality coding equipment and decryption equipment can be constituted using the existing cheap coding network for low bit rates, and a decryption circuit, creation of new coding and Decryption LSI is not needed, but it becomes possible to attain the object cheaply.

[0096] It becomes possible to suppress generating of the quantizing noise greatly generated with the property of input signals, such as Puri Echo, generated in the conventional coding and decryption processing by taking the difference of a decryption processing signal and an input signal also within coding equipment to the 3rd, and taking the configuration of sending the encoded information to decryption equipment.

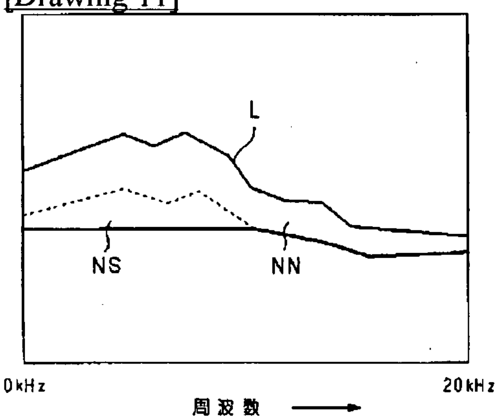
[Translation done.]

DRAWINGS

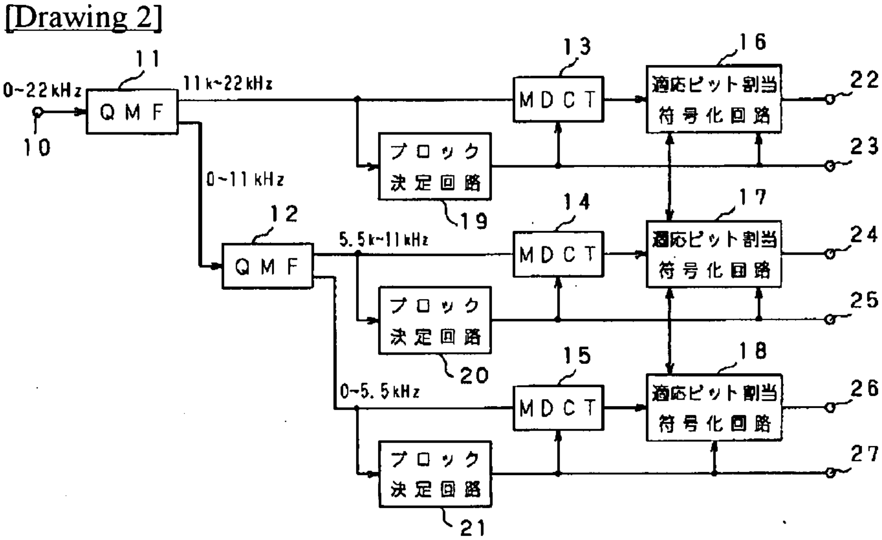
[Drawing 1]



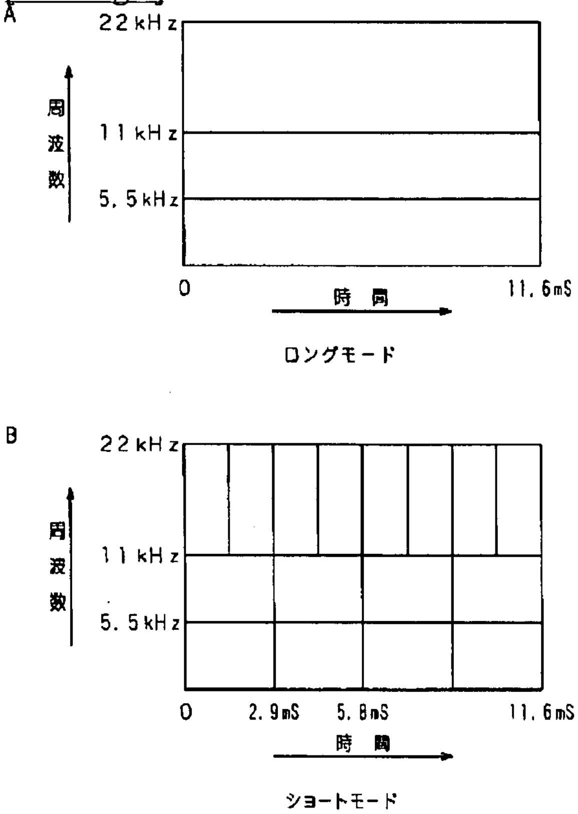
[Drawing 11]



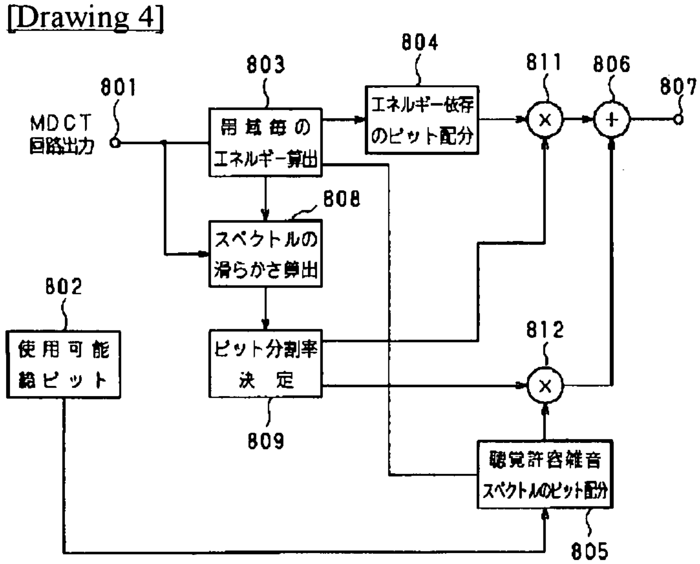
[Drawing 2]



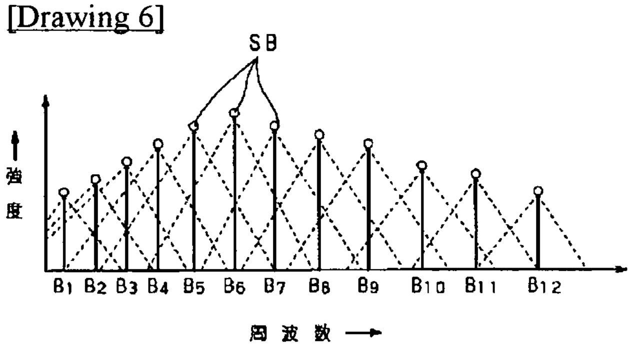
[Drawing 3]



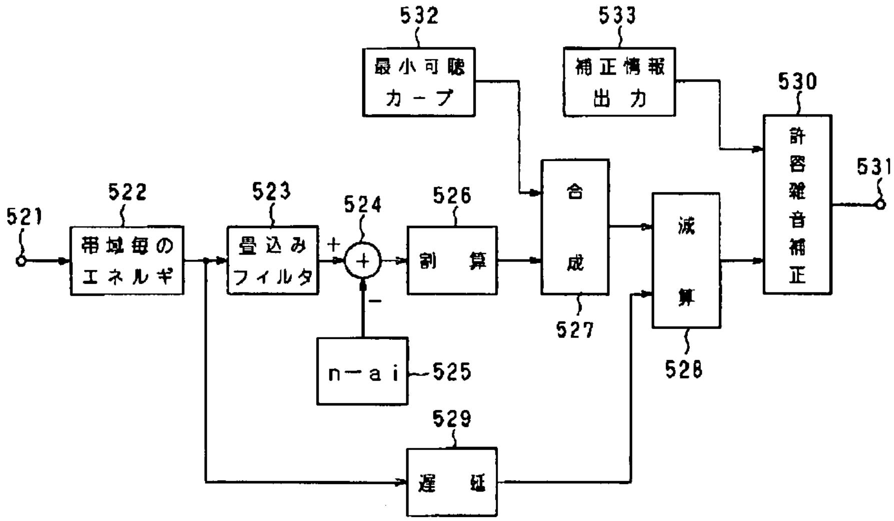
[Drawing 4]



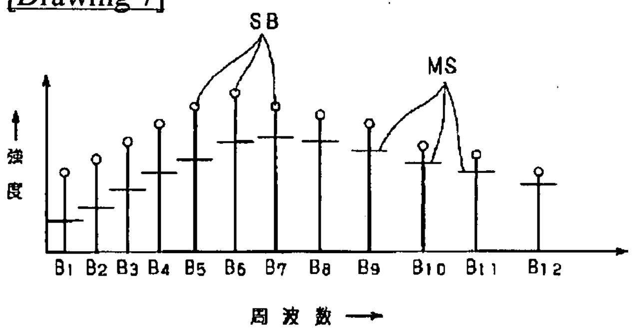
[Drawing 6]



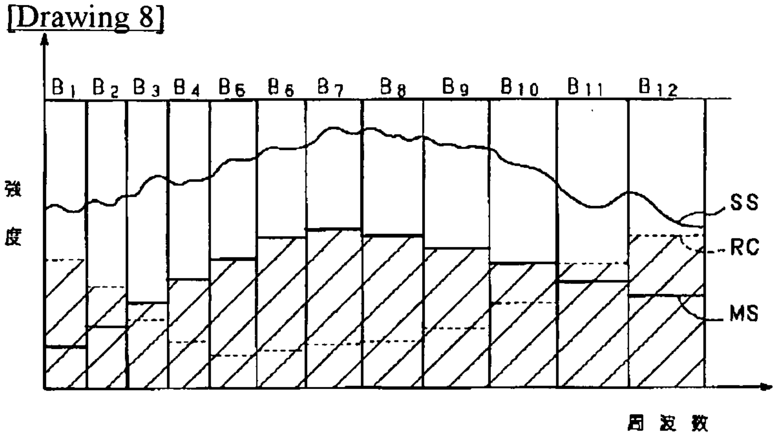
[Drawing 5]



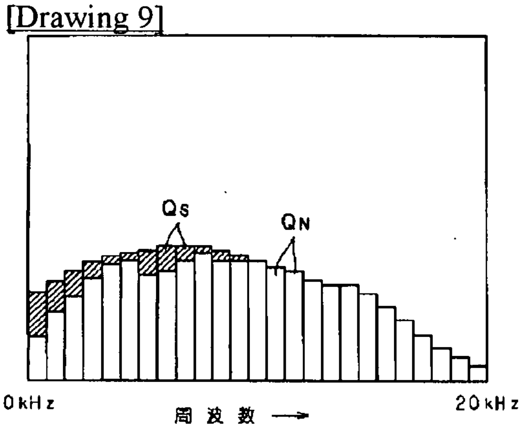
[Drawing 7]



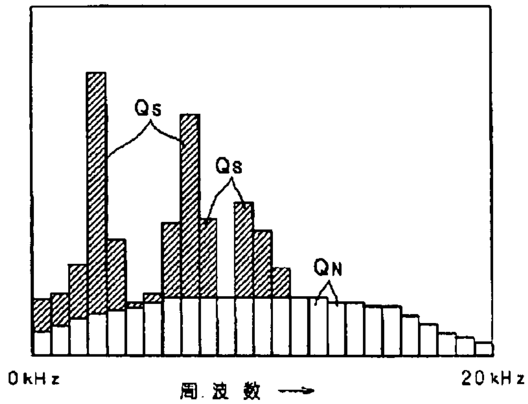
[Drawing 8]



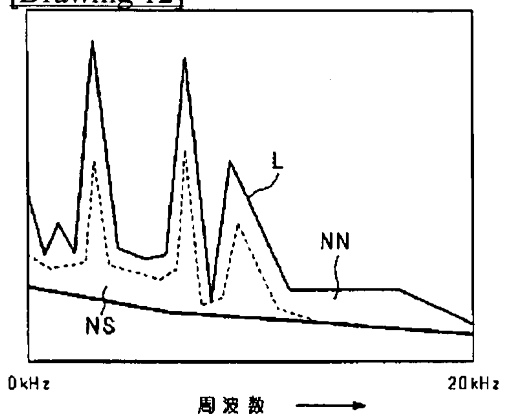
[Drawing 9]



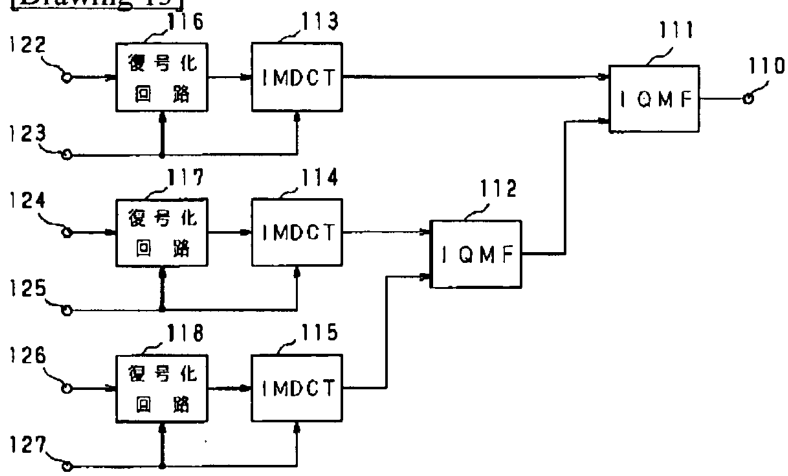
[Drawing 10]



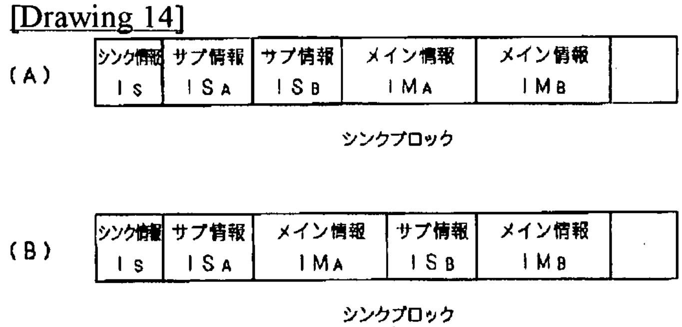
[Drawing 12]



[Drawing 13]



[Drawing 14]



[Translation done.]

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(30)Priority

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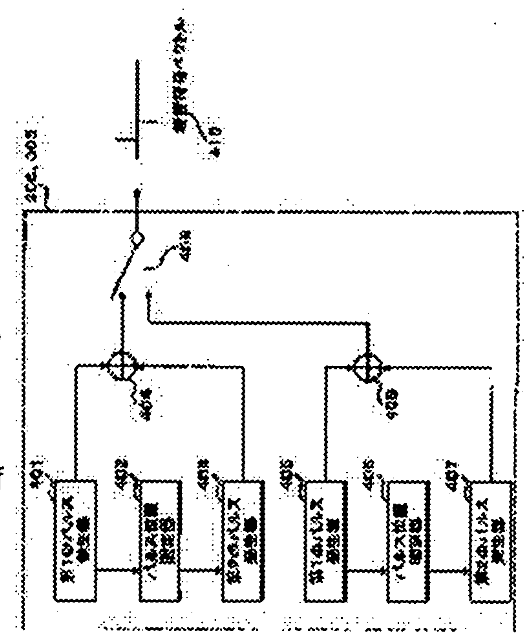
(54) SOUND SOURCE VECTOR GENERATING DEVICE AND VOICE CODING/ DECODING DEVICE

(57)Abstract:

PROBLEM TO BE SOLVED: To improve quality for a silent section and a steady noise section by generating noise code vectors based on first and second pulse positions with two vector generating means, and obtaining a first noise code vector.

SOLUTION: An adder 404 receives a first pulse from a first pulse generator 401 and a second pulse from a second pulse generator 403 and outputs a first noise code vector constituted of two pulses to a selector switch 409. An adder 408 receives a first pulse from a first pulse generator 405 and a second pulse from a second pulse generator 407 and outputs a second noise code vector constituted of two pulses to the selector switch 409. The selector switch 409 selects one of the first noise code vector from the adder 404 and the second noise code vector from the adder 408 and outputs it as a final noise code vector 410.



LEGAL STATUS

[Date of request for examination] 25.10.2006

[Date of sending the examiner's decision of rejection]

* NOTICES *

JPO and INPIT are not responsible for any damages caused by the use of this translation.

1.This document has been translated by computer. So the translation may not reflect the original precisely.
2.**** shows the word which can not be translated.
3.In the drawings, any words are not translated.

DETAILED DESCRIPTION

[Detailed Description of the Invention]
[0001]
[Field of the Invention] This invention relates to the low bit rate voice to digital converter in the migration communication system which encodes and transmits a sound signal, especially a CELP (Code Excited Linear Prediction) mold voice to digital converter which separates into vocal tract information and excitation information, and expresses a sound signal.
[0002]
[Description of the Prior Art] In the field of digital mobile communication or voice storage, speech information is compressed for a deployment of an electric wave or a storage, and the voice to digital converter for encoding in high efficiency is used. The method which used the CELP (Code Excited Linear Prediction: sign excitation linear predictive coding) method as the base is widely put in practical use in inside and a low bit rate especially. the technique of CELP -- M.R.Schroeder and B.S.Atal: -- "Code-Excited Linear Prediction:(CELP) High-quality Speech at Very Low Bit Rates", Proc.ICASSP-85, 25.1.1, pp.937-940, and 1985" It is shown.
[0003] Voice is performed to a certain fixed frame length (5ms - about 50ms), it performs audio linear prediction for every break and frame, and a CELP mold voice coding method encodes the prediction remainder (excitation signal) by the linear prediction for every frame using the adaptation sign vector and noise sign vector which consist of a known wave. An adaptation sign vector is used choosing it from the adaptation sign book which stores the drive sound-source vector generated in the past, and a noise sign vector is used, choosing it from the noise sign book which stores the vector which has the configuration as which the defined number which was prepared beforehand was determined. The vector generated by arranging how many of that pulse in the vector of a random noise sequence or a different location is used for the noise sign vector stored in a noise sign book.
[0004] An algebraic-sign book is in one of the typical things of the noise sign book of the type which arranges several pulses in a different location. About the algebraic-sign book, concrete contents are shown in "ITU-T recommendation G.729" etc.
[0005] The conventional example of the noise sign vector generation machine using an algebraic-sign book is concretely explained below with reference to drawing 40 .
[0006] Drawing 40 is the fundamental block diagram of the noise sign vector generation machine which used the algebraic-sign book. In drawing, the pulse generated from the 1st pulse generator 1 and the 2nd pulse generator 2 is added with an adder 3, and the noise sign vector is generated by standing two pulses to a different location. The example of an algebraic-sign book is shown in drawing 41 and drawing 42 . An example to which drawing 41 stands two pulses into 80 samples, and drawing 42 show an example which stands three pulses into 80 samples, respectively. In addition, in drawing 41 and drawing 42 , the number indicated by the lower part of a table is the number of the combination of the pulse position.
[0007]
[Problem(s) to be Solved by the Invention] However, in the noise sign vector generation machine using the above-mentioned conventional algebraic-sign book, the retrieval location of each sound-source pulse has been independent, and relative location of a certain sound-source pulse and another sound-source pulse is not used. For this reason, when many numbers of bits are needed and a bias is looked at by the configuration of the noise sign vector which should be generated in order to express sufficient pulse position while it is possible to generate the noise sign vector of various configurations, there is a problem that it is not necessarily an efficient sign book. Moreover, in order to reduce the number of bits required for an algebraic-sign book, the technique of reducing a sound-source pulse number can be considered, but since there are few sound-source pulses in this case, there is a problem that the subjective

quality in the silent section or the stationary noise section deteriorates greatly. Moreover, although there is technique of performing the mode change of a sound source in order to improve the subjective quality of the silent section or the stationary noise section, there is a problem when a mode judging error arises.
[0008] It aims at offering the sound-source vector generation equipment and voice coding / decryption equipment which can improve the coding engine performance to non-vocal sound voice or a background noise, this invention being make in view of this point, being able to reduce the size of a noise sign book, being able to improve the quality over the silent section or the stationary noise section, and suppressing quality degradation at the time of a mode judging error moreover.
[0009]
[Means for Solving the Problem] A pulse-position selection means by which the sound-source vector generation equipment of this invention chooses the 1st pulse position from a predetermined pulse-position candidate, A pulse-position decision means to determine the 2nd pulse position which approaches said 1st pulse position on the basis of said 1st pulse position, The configuration possessing at least two vector generation means to generate a noise sign vector based on said 1st and 2nd pulse positions, and a noise sign vector generation means to acquire the 1st noise sign vector generated by said at least two vector generation means is taken.
[0010] According to this configuration, algebraic-sign book size is efficiently reducible by generating the noise sign vector which has at least two pulses which approach mutually.
[0011] The sound-source vector generation equipment of this invention is set in the above-mentioned configuration, and takes the configuration possessing the control means which controls said pulse-position selection means for the pulse position chosen and determined by said pulse-position selection means and said pulse-position decision means not to come out of a transmission frame.
[0012] According to this configuration, the pulse position chosen and determined by the pulse-position selection means and the pulse-position decision means can search in the pulse-position range which does not come out of a transmission frame, and can generate a noise sign vector.
[0013] The sound-source vector generation equipment of this invention possesses the random sign book which stores the 2nd noise sign vector which includes two or more pulses which do not approach mutually in the above-mentioned configuration, and said noise sign vector generation means takes the configuration which generates a noise sign vector from said 1st and 2nd noise sign vectors.
[0014] According to this configuration, the subjective quality over the silent section or the stationary noise section is improvable by using together the random sign book corresponding to non-vocal sound voice or a stationary noise signal with a partial algebraic-sign book.
[0015] The sound-source vector generation equipment of this invention takes the configuration possessing a mode judging means to judge voice mode, and the number control means of pulse-position candidates which makes the number of said predetermined pulse-position candidates fluctuate according to the judged voice mode in the above-mentioned configuration.
[0016] The coding engine performance to non-vocal sound voice or a background noise is improvable, suppressing quality degradation at the time of a mode judging error by changing the use rate of an algebraic-sign book and a random sign book by mode judging according to this configuration.
[0017] The sound-source vector generation equipment of this invention possesses a PAWA calculation means to compute PAWA of an excitation signal, and an average power calculation means to compute average power when the judged voice mode is noise mode, in the above-mentioned configuration, and said number control means of pulse-position candidates makes the number of said predetermined pulse-position candidates fluctuate based on said average power, and takes a configuration.
[0018] According to this configuration, the coding engine performance to non-vocal sound voice or a background noise is improvable, suppressing more efficiently quality degradation at the time of a mode judging error.
[0019] The voice to digital converter of this invention is characterized by having sound-source vector generation equipment of the above-mentioned configuration.
[0020] The voice to digital converter of this invention An excitation vector generation means to generate a new excitation vector from the adaptation sign vector outputted from the adaptation sign book which stored the excitation vector, and the noise sign vector outputted from the partial algebraic-sign book which stored the noise sign vector acquired by sound-source vector generation equipment according to claim 1, The configuration possessing a renewal means of an excitation vector to update the excitation vector stored in the adaptation sign book to said new excitation vector, and a speech synthesis signal generation means to generate a speech synthesis signal using said new excitation vector and the quantized linear-predictive-coding result is taken.

[0021] According to this configuration, by generating the noise sign vector which has at least two pulses which approach mutually, algebraic-sign book size can be reduced efficiently and a voice to digital converter with small bit rate and amount of operations can be realized.
[0022] A sound-source parameter decode means to decode the sound-source parameter with which the voice decryption equipment of this invention includes the index information which specifies the positional information of an adaptation sign vector, and a noise sign vector, An excitation vector generation means to generate an excitation vector using the noise sign vector which has at least two pulses which are acquired from the adaptation sign vector acquired from the positional information of said adaptation sign vector, and said index information, and which approach mutually, The configuration possessing a renewal means of an excitation vector to update the excitation vector stored in the adaptation sign book to said excitation vector, and a speech synthesis signal generation means to generate a speech synthesis signal using said excitation vector and the decrypted quantization linear-predictive-coding result is taken.
[0023] Since the noise sign vector which has at least two pulses which approach mutually is used according to this configuration, algebraic-sign book size can be reduced efficiently and voice decryption equipment with a small bit rate can be realized.
[0024] The voice coding decryption equipment of this invention generates the sound-source vector which consisted of three sound-source pulses, and takes the configuration possessing the partial algebraic-sign book which stores this sound-source vector, a limit means to restrict so that a sound-source vector with at least 1 set of comparatively narrow sound-source pulse separations may be generated among said sound-source vectors, and the random sign book used accommodative according to the size of said partial algebraic-sign book.
[0025] Since according to this configuration a sound-source pulse is set as three pulses and a partial algebraic-sign book is constituted, the high voice coding decryption equipment of fundamentality ability is realizable.
[0026] The voice coding decryption equipment of this invention takes the configuration with which a limit means classifies according to the location (index) of said sound-source pulse in the above-mentioned configuration.
[0027] According to this configuration, since regular sound-source pulse-position retrieval can be performed, the amount of operations which retrieval takes can be held down to necessary minimum.
[0028] In the above-mentioned configuration, the voice coding decryption equipment of this invention divides a partial algebraic-sign book, and takes the configuration only whose part which reduced the size of said partial algebraic-sign book enlarges the rate of a random sign book.
[0029] According to this configuration, even if it changes random sign book size using mode information etc., it is possible to share the index of an intersection, and the effect of errors, such as mode information, can be suppressed.
[0030] In the above-mentioned configuration, the random sign book consists of two or more channels, and the voice coding decryption equipment of this invention takes the configuration which restricts the location of said sound-source pulse as prevents that a sound-source pulse laps between channels.
[0031] Since the orthogonality between the vectors generated from each channel in a sound-source field can be guaranteed according to this configuration, an efficient random sign book can be constituted.
[0032] The voice coding decryption equipment of this invention takes the configuration possessing the algebraic-sign book which stores a sound-source vector, a diffusion pattern generation means to generate a diffusion pattern according to PAWA of the noise section in voice data, and a pattern diffusion means to diffuse the pattern of the sound-source vector outputted from said algebraic-sign book according to said diffusion pattern.
[0033] According to this configuration, since the noise nature of a diffusion pattern is controllable according to noy ZUPAWA, robust voice coding decryption equipment is realizable to noise level.
[0034] In the above-mentioned configuration, the voice coding decryption equipment of this invention generates a diffusion pattern with high noise nature, when average noy ZUPAWA has a large diffusion pattern generation means, and when PAWA is small, it takes the configuration in which a diffusion pattern with low noise nature is generated.
[0035] According to this configuration, when noise level is high, when noise level is low, a cleaner signal can be expressed for a noise-signal.
[0036] The voice coding decryption equipment of this invention takes the configuration in which a diffusion pattern generation means generates a diffusion pattern according to the mode of voice data in the above-mentioned configuration.
[0037] According to this configuration, according to the mode, it also becomes possible to make noise nature of a diffusion pattern into below whenever [middle], and the voice quality in a noise can be improved in the voice section (voiced section).
[0038] The base station equipment of this invention is characterized by having the voice to digital converter of the

above-mentioned configuration. Moreover, the communication terminal of this invention is characterized by having the voice to digital converter of the above-mentioned configuration.

[0039] The pulse-position selection process that the sound-source vector generation method of this invention chooses the 1st pulse position from a predetermined pulse-position candidate, The pulse-position decision process of determining the 2nd pulse position which approaches said 1st pulse position on the basis of said 1st pulse position, At least two vector generation processes which generate a noise sign vector based on said 1st and 2nd pulse positions, and the noise sign vector generation process of acquiring the 1st noise sign vector generated by said at least two vector generation means are provided.

[0040] According to this approach, algebraic-sign book size is efficiently reducible by generating the noise sign vector which has at least two pulses which approach mutually.

[0041]

[Embodiment of the Invention] Using a partial algebraic-sign book generate [the main point of this invention / a noise sign vector], i.e., when at least two among two or more sound-source pulses generated from an algebraic-sign book use the noise sign vector which generates only combination which approaches, algebraic-sign book size is reduced efficiently. Moreover, the subjective quality over the silent section or the stationary noise section is improved using together the random sign book corresponding to non-vocal sound voice or a stationary noise signal with a partial algebraic-sign book, i.e., by storing a sound-source vector effective in the silent section or the stationary noise section. Furthermore, by the mode judging result, suppressing quality degradation at the time of a mode judging error by switching the ratio of partial algebraic-sign book size and the size of the random sign book used together, the coding engine performance to non-vocal sound voice or a background noise is improved, and subjective quality is improved.

[0042] Here, as for the approaching pulse, the distance from a certain pulse says the thing of the pulse which is below 10 sample extent in the digital signal for 1.25 or less ms, i.e., a 8kHz sampling.

[0043] Hereafter, the gestalt of operation of this invention is explained to a detail with reference to an accompanying drawing.

[0044] (Gestalt 1 of operation) Drawing 1 is the block diagram showing a sound signal transmitter and/or a receiver equipped with voice coding and/or decryption equipment concerning this invention.

[0045] In the sound signal transmitter shown in drawing 1 , a sound signal 101 is changed into an electric analog signal by the audio input unit 102, and is outputted to A/D converter 103 by it. An analog sound signal is changed into a digital sound signal by A/D converter 103, and is outputted to a voice to digital converter 104 by it. A voice to digital converter 104 outputs the information which performed voice coding processing and was encoded to the RF modulator 105. In the RF modulator 105, to the encoded sound signal, processing for sending out as electric waves, such as a modulation, magnification, and sign diffusion, is performed, and the encoded sound signal is outputted to the transmitting antenna 1106. Finally an electric wave (RF signal) is sent out from the transmitting antenna 106.

[0046] On the other hand, in a receiver, a receiving antenna 107 receives an electric wave (RF signal). An input signal is sent to the RF demodulator 108. The RF demodulator 108 performs processing for changing electric-wave signals, such as the sign back diffusion of electrons and a recovery, into encoded information, and outputs encoded information to voice decryption equipment 109. Voice decryption equipment 109 performs decode processing of encoded information, and outputs a digital decode sound signal to D/A converter 110. D/A converter 110 changes into an analog decode sound signal the digital decode sound signal outputted from voice decryption equipment 109, and outputs it to an audio output device 111. Finally an audio output device 111 changes and outputs an electric analog decode sound signal to decode voice.

[0047] Next, the noise sign vector generation machine in the sound signal transmitter and/or receiver which have the above-mentioned configuration is explained. Drawing 2 is the block diagram showing the voice to digital converter equipped with the noise sign vector generation machine concerning the gestalt 1 of operation. The voice to digital converter shown in this drawing is equipped with the pretreatment machine 201, the LPC analyzer 202, the LPC quantizer 203, the adaptation sign book 204, a multiplier 205, the partial algebraic-sign book 206, a multiplier 207, an adder 208, the LPC composition filter 209, an adder 210, the acoustic-sense weighting machine 211, and the error minimization machine 212.

[0048] In this noise sign vector generation machine, input voice data is a digital signal acquired by carrying out A/D conversion of the sound signal, and is inputted into the pretreatment machine 201 at every batch time amount (frame). The pretreatment machine 201 improves input voice data in quality subjectively, or performs processing for changing into the signal in the condition of having been suitable for coding, and performs pre-emphasis processing which emphasizes the description of the high-pass filter processing for cutting a dc component for example, or a sound

signal.

[0049] The signal after pretreatment is outputted to the LPC analyzer 202 and an adder 210. The LPC analyzer 202 performs LPC analysis (linear predictive coding) using the signal inputted from the pretreatment machine 201, and outputs obtained LPC (linear predictor coefficients) to the LPC quantizer 203. The LPC quantizer 203 quantizes LPC inputted from the LPC analyzer 202, outputs Quantization LPC to the LPC composition filter 209, and outputs the coded data of Quantization LPC to a decoder side through a transmission line.

[0050] The adaptation sign book 204 is the buffer of the excitation vector (vector outputted from an adder 208) generated in the past, starts an adaptation sign vector from the location specified with the error minimization vessel 212, and outputs it to a multiplier 205. A multiplier 205 multiplies the adaptation sign vector outputted from the adaptation sign book 204 by adaptation sign vector gain, and outputs it to an adder 208. Adaptation sign vector gain is specified with an error minimization vessel. The partial algebraic-sign book 206 is a sign book which has a configuration similar to drawing 4 , drawing 10 , or this which is mentioned later, and outputs the noise sign vector which consists of the pulse which is [several / the location of at least two pulses is close] to a multiplier 207.

[0051] A multiplier 207 multiplies the noise sign vector outputted from the partial algebraic-sign book 206 by noise sign vector gain, and outputs it to an adder 208. By performing vector addition with the adaptation sign vector after the adaptation sign vector gain multiplication outputted from the multiplier 205, and the noise sign vector after the noise sign vector gain multiplication outputted from the multiplier 207, an adder 208 generates an excitation vector and outputs it to the adaptation sign book 204 and the LPC composition filter 209.

[0052] The excitation vector outputted to the adaptation sign book 204 is used when updating the adaptation sign book 204, and the excitation vector outputted to the LPC composition filter 209 is used in order to generate synthesized speech. The LPC composition filter 209 is a linear prediction filter constituted using the quantization LPC outputted from the LPC quantizer 203, drives an LPC composition filter using the excitation vector outputted from the adder 208, and outputs a composite signal to an adder 210.

[0053] the difference (error) of the input sound signal after pretreatment to which the adder 210 was outputted from the pretreatment machine 201, and the composite signal outputted from the LPC composition filter 209 -- a signal is calculated and it outputs to the acoustic-sense weighting machine 211. The acoustic-sense weighting machine 211 performs acoustic-sense-weighting by considering the differential signal outputted from an adder 210 as an input, and outputs it to the error minimization machine 212. The error minimization machine 212 considers the differential signal after acoustic-sense weighting outputted from the acoustic-sense weighting machine 211 as an input. For example, the location which starts an adaptation sign vector from the adaptation sign book 204 so that the square sum may serve as min, The value of the noise sign vector generated from the partial algebraic-sign book 206, the adaptation sign vector gain by which it multiplies with a multiplier 205, and the noise sign vector gain by which it multiplies with a multiplier 207 is adjusted, each is encoded, and it outputs to a decoder side through a transmission line as sound-source parameter coded data.

[0054] Drawing 3 is the block diagram showing voice decryption equipment equipped with the noise sign vector generation machine concerning the gestalt 1 of operation. The voice decryption equipment shown in this drawing is equipped with the LPC decoder 301, the sound-source parameter decoder 302, the adaptation sign book 303, a multiplier 304, the partial algebraic-sign book 305, a multiplier 306, an adder 307, the LPC composition filter 308, and the after-treatment machine 309.

[0055] LPC coded data and sound-source parameter coded data are inputted into the LPC decoder 301 and the sound-source parameter decoder 302 per frame through a transmission line, respectively. The LPC decoder 301 decodes Quantization LPC and outputs it to the LPC composition filter 308. When using Quantization LPC with the after-treatment vessel 309, it is outputted also to the after-treatment machine 309 at coincidence. The sound-source parameter decoder 302 outputs the positional information which starts an adaptation sign vector, adaptation sign vector gain, the index information which specifies a noise sign vector, and noise sign vector gain, respectively to the adaptation sign book 303, a multiplier 304, the partial algebraic-sign book 305, and a multiplier 306.

[0056] The adaptation sign book 303 is the buffer of the excitation vector (vector outputted from an adder 307) generated in the past, starts an adaptation sign vector from the logging location inputted from the sound-source parameter decoder 302, and outputs it to a multiplier 304. A multiplier 304 multiplies the adaptation sign vector outputted from the adaptation sign book 303 by the adaptation sign vector gain inputted from the sound-source parameter decoder 302, and outputs it to an adder 307.

[0057] The partial algebraic-sign book 305 is the same partial algebraic-sign book as what was shown in 206 of drawing 2 which has a configuration similar to drawing 4 , drawing 10 , or this which is mentioned later, and outputs the noise sign vector which consists of the pulse which is [several / the location of at least two pulses specified by the

index inputted from the sound-source parameter decoder 304 is close] to a multiplier 306.
[0058] A multiplier 306 multiplies the noise sign vector outputted from the partial algebraic-sign book by the noise sign vector gain inputted from the sound-source parameter decoder 302, and outputs it to an adder 307. By performing vector addition with the adaptation sign vector after the adaptation sign vector gain multiplication outputted from a multiplier 306, and the noise sign vector after the noise sign vector gain multiplication outputted from the multiplier 306, an adder 307 generates an excitation vector and outputs it to the adaptation sign book 303 and the LPC composition filter 308.
[0059] The excitation vector outputted to the adaptation sign book 303 is used when updating the adaptation sign book 303, and the excitation vector outputted to the LPC composition filter 308 is used in order to generate synthesized speech. The LPC composition filter 308 is a linear prediction filter constituted using the quantization LPC outputted from the LPC decoder 301, drives an LPC composition filter using the excitation vector outputted from the adder 307, and outputs a composite signal to the after-treatment machine 309.
[0060] The processing for improving subjective quality, such as processing for making easy to hear to the synthesized speech outputted from the LPC composition filter 308 the postfilter processing and the steady background noise which consist of formant emphasis processing, pitch emphasis processing, spectrum inclination amendment processing, etc., is carried out, and the after-treatment machine 309 outputs as decode voice data.
[0061] Next, the noise sign vector generation machine concerning this invention is explained to a detail. Drawing 4 is the block diagram showing the configuration of the noise sign vector generation equipment concerning the gestalt 1 of operation of this invention.
[0062] The 1st pulse generator 401 stands the 1st pulse to one of the location candidates set beforehand as shown in the column of the pulse number 1 of drawing 5 (a), and outputs it to an adder 404. Moreover, the 1st pulse generator 401 outputs at coincidence the positional information which stood the 1st pulse to the pulse-position limited machine 402. The pulse-position limited machine 402 inputs the 1st pulse position from the 1st pulse generator 401, and determines the location candidate of the 2nd pulse on the basis of the location.
[0063] The location candidate of the 2nd pulse is expressed with the relative expression from the location (= P1) of the 1st pulse as shown in the column of the pulse number 2 of drawing 5 (a). The pulse-position limited machine 402 outputs the location candidate of the 2nd pulse to the 2nd pulse generator 403. The 2nd pulse generator 403 stands the 2nd pulse to one of the location candidates of the 2nd pulse inputted from the pulse-position limited machine 402, and outputs it to an adder 404.
[0064] An adder 404 outputs the 1st noise sign vector which inputs the 1st pulse outputted from the 1st pulse generator 401, and the 2nd pulse outputted from the 2nd pulse generator 403, and consists of two pulses to a circuit changing switch 409.
[0065] On the other hand, the 2nd pulse generator 407 stands the 2nd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 2 of drawing 5 (b), and outputs it to an adder 408. Moreover, the 2nd pulse generator 407 outputs at coincidence the positional information which stood the 2nd pulse to the pulse-position limited machine 406. The pulse-position limited machine 406 inputs the 2nd pulse position from the 2nd pulse generator 407, and determines the location candidate of the 1st pulse on the basis of the location.
[0066] The location candidate of the 1st pulse is expressed with the relative expression from the location (= P2) of the 2nd pulse as shown in the column of the pulse number 1 of drawing 5 (b). The pulse-position limited machine 406 outputs the location candidate of the 1st pulse to the 1st pulse generator 405. The 1st pulse generator 405 stands the 1st pulse to one of the location candidates of the 1st pulse inputted from the pulse-position limited machine 406, and outputs it to an adder 408.
[0067] An adder 408 outputs the 2nd noise sign vector which inputs the 1st pulse outputted from the 1st pulse generator 405, and the 2nd pulse outputted from the 2nd pulse generator 407, and consists of two pulses to a circuit changing switch 409.
[0068] A circuit changing switch 409 chooses one of the 1st noise sign vector outputted from an adder 404, and the 2nd noise sign vector outputted from an adder 408, and outputs it as a final noise sign vector 410. This selection is specified by control from the outside.
[0069] In addition, when the pulse which expresses one side absolutely as mentioned above between two pulses in a location, and is absolutely expressed in a location when another side is expressed as mentioned above in a relative position is near a frame tail, the pulse expressed in a relative position may overflow out of a frame. For this reason, in an actual search algorithm, it is possible to divide into three kinds of retrieval location patterns (a-c), and to search, as only the part which the combination to protrude produces is used as another pattern and shown in drawing 5 . Drawing 5 makes frame length 80 samples (0-79), and an example in the case of standing two pulses into one frame is

shown. From the sign book shown in drawing 5 , a part of total entry of a noise sign vector generable from the conventional algebraic-sign book shown in drawing 40 is generable. Suppose that the algebraic-sign book of this invention as shown in drawing 5 is called a partial algebraic-sign book in this semantics.

[0070] With reference to drawing 6 - drawing 8 , it explains below that processing of the noise sign vector generation method (the coding approach, the noise sign book retrieval approach) in the gestalt of the above-mentioned implementation using the sign book of drawing 5 flows. By drawing 6 , the polarity (+, -) of a pulse shows concretely the case which carries out a thing assumption and encodes only the location of a pulse where it encodes separately.

[0071] First, in step (it abbreviates to ST hereafter) 601, initialization of loop variable i, the error function maximum Max, Index idx, the output index index, the 1st pulse position position1, and the 2nd pulse position position2 is performed.

[0072] Loop variable i is used as a loop variable of the pulse expressed absolutely in a location here, and initial value is 0. The error function maximum Max is initialized by the minimum value (for example, "-10^32") which can be expressed, and it is used in order to maximize the error performance index calculated by the retrieval loop formation. Index idx is an index given to each of the code vector generated by this noise sign vector generation method, initial value is 0, and whenever it changes one location of a pulse, the increment of it is carried out. The location of the 1st pulse where the index of a noise sign vector with which index is finally outputted, and position1 are finally determined, and position2 are the locations of the 2nd pulse finally determined.

[0073] Next, the 1st pulse position (p1) is set to pos1a [j] in ST602. pos1a[ ] is a location (0, 2, .., 72) shown in the column of the pulse number 1 of drawing 5 (a). Here, the 1st pulse is a pulse expressed absolutely in a location.

[0074] Next, initialization of loop variable j is performed in ST PU 603. Loop variable j is the loop variable of the pulse expressed in a relative position, and initial value is 0. Here, the 2nd pulse is expressed in a relative position.

[0075] Next, the 2nd pulse position (p2) is set to p1+pos2a [j] in ST604. p1 is the 1st pulse position already set in ST602, and is pos2a[4] = {1, 3, 5, 7}. The size (the total number of entries of a noise sign vector) of a partial algebraic-sign book can be reduced by reducing the number of elements of pos2a[]. In this case, it is necessary to change the contents of drawing 5 (c) according to the reduced number. Moreover, it is also the same as when increasing.

[0076] Next, in ST605, the error performance index E at the time of standing a pulse to the two set pulse positions is calculated. An error performance index is for evaluating the error of the vector used as a target, and the vector compounded from a noise sign vector, for example, the following formula (1) is used. In addition, when orthogonalizing a noise sign vector to an adaptation sign vector so that it may generally be well used with a CELP encoder, the formula which transformed the formula (1) will be used. When the value of a formula (1) becomes max, an error with the synthetic vector acquired by driving a synthetic filter by the vector and noise sign vector which are used as the target serves as min.

[Equation 1]

$$\frac{(x'Hci)^2}{ci'H'Hci}$$

x:ターゲットベクトル
H:合成フィルタのインパルス畳み込み行列
c:雑音符号ベクトル(iはインデックス番号))

[0077] Next, in ST606, it judges whether the value of the error performance index E is over the error performance-index maximum Max. If E value is over Maximum Max, and it progresses to ST607 and has not exceeded, ST607 will be skipped and it will progress to ST608.

[0078] In ST607, renewal of position1 and position2 is performed with index and Max. That is, the error performance-index maximum Max is updated to the error performance index E calculated by ST605, index is updated to idx, position1 is updated in the location p1 of the 1st pulse, and position2 is updated in the location p2 of the 2nd pulse.

[0079] Next, in ST608, loop variable j and the index number idx are incremented, respectively. By incrementing loop variable j, the location of the 2nd pulse will be moved and the noise sign vector of the following index number will be evaluated.

[0080] Next, in ST609, loop variable j checks that it is under total NUM2a of the location candidate of the 2nd pulse. By the partial algebraic-sign book shown in drawing 5 , it is NUM2a=4. When loop variable j is under NUM2a, in order to repeat the loop formation of j, it returns to ST604. If loop variable j has reached NUM2a, it will end and the loop formation of j will progress to ST610.

[0081] The increment of loop variable i is performed in ST610. By incrementing loop variable i, the location of the 1st pulse will be moved and the noise sign vector of the following index number will be evaluated.
[0082] Next, in ST611, loop variable i checks that it is under total NUM1a of the location candidate of the 1st pulse. By the partial algebraic-sign book shown in drawing 5 , it is NUM1a=37. When loop variable i is under NUM1a, in order to repeat the loop formation of i, it returns to ST602. If loop variable i has reached NUM1a, it will end and the loop formation of i will progress to ST701 of drawing 7 . When it progresses to ST612, retrieval of drawing 5 (a) is ended and the retrieval loop formation of drawing 5 (b) is started.
[0083] Next, in ST701, loop variable i is cleared and it is set to 0. The 2nd pulse position (p2) is set to pos2b [i] in ST702. pos2b[ ] is a location (1, 3, .., 61) shown in the column of the pulse number 2 of drawing 5 (b). Here, the 2nd pulse is a pulse expressed absolutely in a location.
[0084] Next, initialization of loop variable j is performed in ST703. Loop variable j is the loop variable of the pulse expressed in a relative position, and initial value is 0. Here, the 1st pulse is expressed in a relative position.
[0085] Next, the 1st pulse position (p1) is set to p2+pos1b [j] in ST704. p2 is the 2nd pulse position and pos1b[4] = {1, 3, 5, 7} which have already been set in ST702. The size (the total number of entries of a noise sign vector) of a partial algebraic-sign book can be reduced by reducing the number of elements of pos1b[ ]. In this case, it is necessary to change the contents of drawing 5 (c) according to the reduced number. Moreover, it is also the same as when increasing the number of elements of pos1b[ ].
[0086] Next, in ST705, the error performance index E at the time of standing a pulse to the two set pulse positions is calculated. A formula as an error performance index been for evaluating the error of the vector used as a target and the vector compounded from a noise sign vector, for example, shown in a formula (1) is used. In addition, when orthogonalizing a noise sign vector to an adaptation sign vector so that it may generally be well used with a CELP encoder, the formula which transformed the formula (1) will be used. When the value of a formula (1) becomes max, an error with the synthetic vector acquired by driving a synthetic filter by the vector and noise sign vector which are used as the target serves as min.
[0087] Next, in ST706, it judges whether the value of the error performance index E is over the error performance-index maximum Max. If E value is over Maximum Max, and it progresses to ST707 and has not exceeded, ST707 is skipped and it progresses to ST708.
[0088] In ST707, renewal of position1 and position2 is performed with index and Max. That is, the error performance-index maximum Max is updated to the error performance index E calculated by ST705, index is updated to idx, position1 is updated in the location p1 of the 1st pulse, and position2 is updated in the location p2 of the 2nd pulse.
[0089] Next, in ST708, loop variable j and the index number idx are incremented, respectively. By incrementing loop variable j, the location of the 1st pulse will be moved and the noise sign vector of the following index number will be evaluated.
[0090] Next, in ST709, loop variable j checks that it is under total NUM1b of the location candidate of the 1st pulse. By the partial algebraic-sign book shown in drawing 5 , it is NUM1b=4. When loop variable j is under NUM1b, in order to repeat the loop formation of j, it returns to ST704. If loop variable j has reached NUM1b, it will end and the loop formation of j will progress to ST710.
[0091] The increment of loop variable i is performed in ST701. By incrementing loop variable i, the location of the 2nd pulse will be moved and the noise sign vector of the following index number will be evaluated.
[0092] Next, in ST711, loop variable i checks that it is under total NUM2b of the location candidate of the 2nd pulse. By the partial algebraic-sign book shown in drawing 5 , it is NUM2b=36. When loop variable i is under NUM2b, in order to repeat the loop formation of i, it returns to ST702. If loop variable i has reached NUM2b, it will end and the loop formation of i will progress to ST801 of drawing 8 . When it progresses to ST801, retrieval of drawing 5 (b) is ended and the retrieval loop formation of drawing 5 (c) is started.
[0093] In ST801, loop variable i is cleared and it is set to 0. Next, the 1st pulse position (p1) is set to pos1c [i] in ST802. pos1c[ ] is a location (74, 76, 78) shown in the column of the pulse number 1 of drawing 5 (c). here -- the -- the pulse of both [2nd] one is expressed absolutely in a location.
[0094] Next, initialization of loop variable j is performed in ST803. Loop variable j is the loop variable of the 2nd pulse, and initial value is 0.
[0095] Next, the 2nd pulse position (p2) is set to pos2c [j] in ST804. pos2c[ ] is a location {73, 75, 77, 79} shown in the column of the pulse number 2 of drawing 5 (c).
[0096] Next, in ST805, the error function E at the time of standing a pulse to the two set pulse positions is calculated. A formula as an error function been for evaluating the error of the vector used as a target and the vector compounded

from a noise sign vector, for example, shown in a formula (1) is used. In addition, when orthogonalizing a noise sign vector to an adaptation sign vector so that it may generally be well used with a CELP encoder, the formula which transformed the formula (1) will be used. When the value of a formula (1) becomes max, an error with the synthetic vector acquired by driving a synthetic filter by the vector and noise sign vector which are used as the target serves as min.

[0097] Next, in ST806, it judges whether the value of the error performance index E is over the error performance-index maximum Max. If it has exceeded, and progresses to ST807 and has not exceeded, ST807 is skipped and it progresses to ST808. In ST807, renewal of position1 and position2 is performed with index and Max. That is, the error performance-index maximum Max is updated to the error performance index E calculated by ST805, index is updated to idx, position1 is updated in the location p1 of the 1st pulse, and position2 is updated in the location p2 of the 2nd pulse.

[0098] Next, in ST808, loop variable j and the index number idx are incremented, respectively. By incrementing loop variable j, the location of the 2nd pulse will be moved and the noise sign vector of the following index number will be evaluated.

[0099] Next, in ST809, loop variable j checks that it is under total NUM2c of the location candidate of the 2nd pulse. By the partial algebraic-sign book shown in drawing 5 , it is NUM2c=4. When loop variable j is under NUM2c, in order to repeat the loop formation of j, it returns to ST804. If loop variable j has reached NUM2c, it will end and the loop formation of j will progress to ST810.

[0100] The increment of loop variable i is performed in ST810. By incrementing loop variable i, the location of the 1st pulse will be moved and the noise sign vector of the following index number will be evaluated.

[0101] Next, in ST811, loop variable i checks that it is under total NUM1c of the location candidate of the 1st pulse. By the partial algebraic-sign book shown in drawing 5 , it is NUM1c=3. When loop variable i is under NUM1c, in order to repeat the loop formation of i, it returns to ST802. If loop variable i has reached NUM1c, it will end and the loop formation of i will progress to ST812. When it progresses to ST812, it ends and all retrieval ends retrieval of drawing 5 (c).

[0102] Finally, in ST812, index which it is as a result of retrieval is outputted. Although it is not necessary to output the two pulse positions position1 and position2 corresponding to index, it can be used for local decode. In addition, since the polarity (are they + or -?) of each pulse can be beforehand determined by doubling with the vector xH in a formula (1) (only the time of the correlation of xH and c in a formula (1) being forward is considered), it is omitted with the gestalt of the above-mentioned implementation.

[0103] With reference to drawing 9 , the flow of processing of the noise sign vector generation method (the decryption approach) in the gestalt of the above-mentioned implementation using the sign book of drawing 5 is explained below. By drawing 9 , the polarity (+, -) of a pulse shows concretely the case which carries out a thing assumption and decrypts only the location of a pulse where it is decrypted separately.

[0104] First, in ST901, the index index of the noise sign vector received from the encoder confirms whether to be less than one IDX. IDX1 is the sign book size of the part of (a) in the sign book of drawing 5 , and is the value of idx in the time in ST601 of drawing 6 . It is more specifically IDX1=32x4=128. If index is less than one IDX, since the two pulse positions are parts expressed by drawing 5 (a), it progresses to ST602. In order to check further since it becomes drawing 5 (b) or the part of (c) when index(es) are one or more IDX(s), it progresses to ST905.

[0105] It asks for the quotient idx1 which broke index by Num2a in ST902. idx1 serves as an index number of the 1st pulse. In ST902, int() is a function which asks for the integer part in ( ).

[0106] Next, in ST903, idx2 is calculated just because it broke index by Num2a. idx2 serves as an index number of the 2nd pulse.

[0107] Next, in ST904, the location position2 of the 2nd pulse is determined using the sign book of drawing 5 (a) using idx2 asked for the location position1 of the 1st pulse using idx1 calculated by ST902 by ST903, respectively. position1 and position2 which were determined are used by ST914.

[0108] When index(es) are one or more IDX(s) in ST901, it progresses to ST905. In ST905, index confirms whether to be less than two IDX. IDX2 is the sign book size which doubled the part of (a) in the sign book of drawing 5 , and the part of (b), and is the value of idx in the time in ST801 of drawing 6 . It is more specifically IDX2=32x4+31x4=252. If index is less than two IDX, since the two pulse positions are parts expressed by drawing 5 (b), it progresses to ST906. Since it is the part expressed by drawing 5 (c) when index(es) are two or more IDX(s), it progresses to ST910.

[0109] In ST906, IDX1 is subtracted from index and it progresses to ST907. It asks for the quotient idx2 which broke index after IDX1 subtraction by Num1b in ST907. This idx2 serves as an index number of the 2nd pulse. In ST907,

int ( ) is a function which asks for the integer part in ( ).
[0110] Next, in ST908, idx1 is calculated just because it broke index after IDX1 subtraction by Num1b. This idx1 serves as an index number of the 1st pulse.
[0111] Next, in ST909, the location position1 of the 1st pulse is determined using the sign book of drawing 5 (b) using idx1 asked for the location position2 of the 2nd pulse using idx2 calculated by ST907 by ST908, respectively. position1 and position2 which were determined are used by ST914.
[0112] When index(es) are two or more IDX(s) in ST905, it progresses to ST910. In ST910, IDX2 is subtracted from index and it progresses to ST911. It asks for the quotient idx1 which broke index after IDX2 subtraction by Num2c in ST911. This idx1 serves as an index number of the 1st pulse. In ST911, int ( ) is a function which asks for the integer part in ( ).
[0113] Next, in ST912, idx2 is calculated just because it broke index after IDX2 subtraction by Num2c. This idx2 serves as an index number of the 2nd pulse.
[0114] Next, in ST913, the location position2 of the 2nd pulse is determined using the sign book of drawing 5 (c) using idx2 asked for the location position1 of the 1st pulse using idx1 calculated by ST911 by ST912, respectively. position1 and position2 which were determined are used by ST914.
[0115] In ST914, noise sign vector code[ ] is generated using the location position1 of the 1st pulse, and the location position2 of the 2nd pulse. That is, the vector which is 0 is generated except code [position1] and code [position2]. code "position1" and code "position2" are set to the polarities [sign / sign and / +1] 1 decoded separately or 1 by two (sign1 and sign2 take the value of +1 or 1). It is the noise sign vector by which code[ ] is decoded.
[0116] Next, the example of a configuration of the partial algebraic-sign book whose pulse number is three is shown in drawing 10 .
[0117] The example of a configuration in drawing 10 takes the configuration which limits a pulse retrieval location so that at least two of three may be arranged in the location which approached. The sign book corresponding to this configuration is shown in drawing 11 .
[0118] Explanation is added to below using drawing 10 . The 1st pulse generator 1001 stands the 1st pulse to one of the location candidates set beforehand as shown in the column of the pulse number 1 of drawing 11 (a), and outputs it to an adder 1005. Moreover, the 1st pulse generator 1001 outputs at coincidence the positional information which stood the 1st pulse to the pulse-position limited machine 1002. The pulse-position limited machine 1002 inputs the positional information of the 1st pulse from the 1st pulse generator 1001, and determines the location candidate of the 2nd pulse on the basis of the location. The location candidate of the 2nd pulse is expressed with the relative expression from the location (= P1) of the 1st pulse as shown in the column of the pulse number 2 of drawing 11 (a).
[0119] The pulse-position limited machine 1002 outputs the candidate of the 2nd pulse position to the 2nd pulse generator 1003. The 2nd pulse generator 1003 stands the 2nd pulse to one of the location candidates of the 2nd pulse inputted from the pulse-position limited machine 1002, and outputs it to an adder 1005. The 3rd pulse generator 1004 stands the 3rd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 3 of drawing 11 (a), and outputs it to an adder 1005. An adder 1005 performs vector addition of a total of three impulse vectors outputted from each pulse generator of 1001, 1003, and 1004, and outputs the noise sign vector which consists of three pulses to a change-over switch 1031.
[0120] The 1st pulse generator 1006 stands the 1st pulse to one of the location candidates set beforehand as shown in the column of the pulse number 1 of drawing 11 (d), and outputs it to an adder 1010. Moreover, the 1st pulse generator 1006 outputs at coincidence the positional information which stood the 1st pulse to the pulse-position limited machine 1007. The pulse-position limited machine 1007 inputs the positional information of the 1st pulse from the 1st pulse generator 1006, and determines the location candidate of the 3rd pulse on the basis of the location. The location candidate of the 3rd pulse is expressed with the relative expression from the location (= P1) of the 1st pulse as shown in the column of the pulse number 3 of drawing 11 (d).
[0121] The pulse-position limited machine 1007 outputs the candidate of the 3rd pulse position to the 3rd pulse generator 1008. The 3rd pulse generator 1008 stands the 3rd pulse to one of the location candidates of the 3rd pulse inputted from the pulse-position limited machine 1007, and outputs it to an adder 1010. The 2nd pulse generator 1009 stands the 2nd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 2 of drawing 11 (d), and outputs it to an adder 1010. An adder 1010 performs vector addition of a total of three impulse vectors outputted from each pulse generator of 1006, 1008, and 1009, and outputs the noise sign vector which consists of three pulses to a change-over switch 1031.
[0122] The 3rd pulse generator 1011 stands the 3rd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 3 of drawing 11 (b), and outputs it to an adder 1015. The 2nd pulse generator 1012

stands the 2nd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 2 of drawing 11 (b), and outputs it to an adder 1015. Moreover, the 2nd pulse generator 1012 outputs at coincidence the location which stood the 2nd pulse to the pulse-position limited machine 1013. The pulse-position limited machine 1013 inputs the location of the 2nd pulse from the 2nd pulse generator 1012, and determines the location candidate of the 1st pulse on the basis of the location. The location candidate of the 1st pulse is expressed with the relative expression from the location (= P2) of the 2nd pulse as shown in the column of the pulse number 1 of drawing 11 (b).

[0123] The pulse-position limited machine 1013 outputs the location candidate of the 1st pulse to the 1st pulse generator 1014. The 1st pulse generator 1014 stands the 1st pulse to one of the location candidates of the 1st pulse inputted from the pulse-position limited machine 1013, and outputs it to an adder 1015. An adder 1015 performs vector addition of a total of three impulse vectors outputted from each pulse generator of 1011, 1012, and 1014, and outputs the noise sign vector which consists of three pulses to a circuit changing switch 1031.

[0124] The 1st pulse generator 1016 stands the 1st pulse to one of the location candidates set beforehand as shown in the column of the pulse number 1 of drawing 11 (g), and outputs it to an adder 1020. The 2nd pulse generator 1017 stands the 2nd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 2 of drawing 11 (g), and outputs it to an adder 1020. Moreover, the 2nd pulse generator 1017 outputs at coincidence the location which stood the 2nd pulse to the pulse-position limited machine 1018. The pulse-position limited machine 1018 inputs the location of the 2nd pulse from the 2nd pulse generator 1017, and determines the location candidate of the 3rd pulse on the basis of the location. The location candidate of the 3rd pulse is expressed with the relative expression from the location (= P2) of the 2nd pulse as shown in the column of the pulse number 3 of drawing 11 (g).

[0125] The pulse-position limited machine 1018 outputs the location candidate of the 3rd pulse to the 3rd pulse generator 1019. The 3rd pulse generator 1019 stands the 3rd pulse to one of the location candidates of the 3rd pulse inputted from the pulse-position limited machine 1018, and outputs it to an adder 1020. An adder 1020 performs vector addition of a total of three impulse vectors outputted from each pulse generator of 1016, 1017, and 1019, and outputs the noise sign vector which consists of three pulses to a change-over switch 1031.

[0126] The 2nd pulse generator 1021 stands the 2nd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 2 of drawing 11 (e), and outputs it to an adder 1025. The 3rd pulse generator 1024 stands the 3rd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 3 of drawing 11 (e), and outputs it to an adder 1025. Moreover, the 3rd pulse generator 1024 outputs at coincidence the location which stood the 3rd pulse to the pulse-position limited machine 1023. The pulse-position limited machine 1023 inputs the location of the 3rd pulse from the 3rd pulse generator 1024, and determines the location candidate of the 1st pulse on the basis of the location. The location candidate of the 1st pulse is expressed with the relative expression from the location (= P3) of the 3rd pulse as shown in the column of the pulse number 1 of drawing 11 (e).

[0127] The pulse-position limited machine 1023 outputs the location candidate of the 1st pulse to the 1st pulse generator 1022. The 1st pulse generator 1022 stands the 1st pulse to one of the location candidates of the 1st pulse inputted from the pulse-position limited machine 1023, and outputs it to an adder 1025. An adder 1025 performs vector addition of a total of three impulse vectors outputted from each pulse generator of 1021, 1022, and 1024, and outputs the noise sign vector which consists of three pulses to a change-over switch 1031.

[0128] The 1st pulse generator 1026 stands the 1st pulse to one of the location candidates set beforehand as shown in the column of the pulse number 1 of drawing 11 (h), and outputs it to an adder 1030. The 3rd pulse generator 1029 stands the 3rd pulse to one of the location candidates set beforehand as shown in the column of the pulse number 3 of drawing 11 (h), and outputs it to an adder 1030. Moreover, the 3rd pulse generator 1029 outputs at coincidence the location which stood the 3rd pulse to the pulse-position limited machine 1028. The pulse-position limited machine 1028 inputs the location of the 3rd pulse from the 3rd pulse generator 1029, and determines the location candidate of the 2nd pulse on the basis of the location. The location candidate of the 2nd pulse is expressed with the relative expression from the location (= P3) of the 3rd pulse as shown in the column of the pulse number 2 of drawing 11 (h).

[0129] The pulse-position limited machine 1028 outputs the location candidate of the 2nd pulse to the 2nd pulse generator 1027. The 2nd pulse generator 1027 stands the 2nd pulse to one of the location candidates of the 2nd pulse inputted from the pulse-position limited machine 1028, and outputs it to an adder 1030. An adder 1030 performs vector addition of a total of three impulse vectors outputted from each pulse generator of 1026, 1027, and 1029, and outputs the noise sign vector which consists of three pulses to a change-over switch 1031.

[0130] A change-over switch 1031 chooses one from a total of six kinds of noise sign vectors inputted from each adder of 1005, 1010, 1015, 1020, 1025, and 1030, and outputs the noise sign vector 1032. This selection is specified by control from the outside.

[0131] In addition, although drawing 5 (c), drawing 11 (c), (f), and (i) are prepared supposing the case where the

pulse expressed in a relative position overflows a frame in drawing 5 and drawing 11 Since the range of the location candidate of the pulse expressed absolutely in a location is partial ahead of the frame, when it is not possible that the pulse expressed in a relative position overflows a frame, these parts (drawing 5 (c) etc.) can be omitted.

[0132] (Gestalt 2 of operation) Drawing 12 is the block diagram showing the voice to digital converter equipped with the noise sign vector generation machine concerning the gestalt 2 of operation. The voice to digital converter shown in this drawing is equipped with the pretreatment machine 1201, the LPC analyzer 1202, the LPC quantizer 1203, the adaptation sign book 1204, a multiplier 1205, the noise sign book 1206 that consists of a partial algebraic-sign book and a random sign book, a multiplier 1207, an adder 1208, the LPC composition filter 1209, an adder 1210, the acoustic-sense weighting machine 1211, and the error minimization machine 1212.

[0133] In this voice to digital converter, input voice data is a digital signal acquired by carrying out A/D conversion of the sound signal, and is inputted into the pretreatment machine 1201 at every batch time amount (frame). The pretreatment machine 1201 improves input voice data in quality subjectively, or performs processing for changing into the signal in the condition of having been suitable for coding, and performs pre-emphasis processing which emphasizes the description of the high-pass filter processing for cutting a dc component for example, or a sound signal.

[0134] The signal after pretreatment is outputted to the LPC analyzer 1202 and an adder 1210. The LPC analyzer 1202 performs LPC analysis (linear predictive coding) using the signal inputted from the pretreatment machine 1201, and outputs obtained LPC (linear predictor coefficients) to the LPC quantizer 1203. The LPC quantizer 1203 quantizes LPC inputted from the LPC analyzer 1202, outputs Quantization LPC to the LPC composition filter 1209, and outputs the coded data of Quantization LPC to a decoder side through a transmission line.

[0135] The adaptation sign book 1204 is the buffer of the excitation vector (vector outputted from an adder 1208) generated in the past, starts an adaptation sign vector from the location specified with the error minimization vessel 1212, and outputs it to a multiplier 1205. A multiplier 1205 multiplies the adaptation sign vector outputted from the adaptation sign book 1204 by adaptation sign vector gain, and outputs it to an adder 1208. Adaptation sign vector gain is specified with an error minimization vessel.

[0136] The noise sign book 1206 which consists of a partial algebraic-sign book and a random sign book is a sign book with the configuration shown in drawing 14 mentioned later, and outputs either the noise sign vector which consists of the pulse which is [several / the location of at least two pulses is close], or the noise sign vector of about 90% or less of rates of sparse (the measurement size of the amplitude zero to the measurement size of the whole frame comparatively) to a multiplier 1207.

[0137] A multiplier 1207 multiplies the noise sign vector outputted from the noise sign book 1206 which consists of a partial algebraic-sign book and a random sign book by noise sign vector gain, and outputs it to an adder 1208. By performing vector addition with the adaptation sign vector after the adaptation sign vector gain multiplication outputted from the multiplier 1205, and the noise sign vector after the noise sign vector gain multiplication outputted from the multiplier 1207, an adder 1208 generates an excitation vector and outputs it to the adaptation sign book 1204 and the LPC composition filter 1209.

[0138] The excitation vector outputted to the adaptation sign book 1204 is used for updating the adaptation sign book 1204, and the excitation vector outputted to the LPC composition filter 1209 is used in order to generate synthesized speech. The LPC composition filter 1209 is a linear prediction filter constituted using the quantization LPC outputted from the LPC quantizer 1203, drives an LPC composition filter using the excitation vector outputted from the adder 1208, and outputs a composite signal to an adder 1210. the difference (error) of the input sound signal after pretreatment to which the adder 1210 was outputted from the pretreatment machine 1201, and the composite signal outputted from the LPC composition filter 1209 -- a signal is calculated and it outputs to the acoustic-sense weighting machine 1211.

[0139] The acoustic-sense weighting machine 1211 performs acoustic-sense-weighting by considering the differential signal outputted from an adder 1210 as an input, and outputs it to the error minimization machine 1212. The error minimization machine 1212 considers the differential signal after acoustic-sense weighting outputted from the acoustic-sense weighting machine 1211 as an input. For example, the square sum So that it may become min an adaptation sign vector from the adaptation sign book 1204 The value of the noise sign vector generated from the noise sign book 1206 which consists of the location and partial algebraic-sign book to cut down, and a random sign book, the adaptation sign vector gain by which it multiplies with a multiplier 1205, and the noise sign vector gain by which it multiplies with a multiplier 1207 is adjusted. Each is encoded and it outputs to a decoder side through a transmission line as sound-source parameter coded data 1214.

[0140] Drawing 13 is the block diagram showing voice decryption equipment equipped with the noise sign vector

generation machine concerning the gestalt 2 of operation. The voice decryption equipment shown in this drawing is equipped with the LPC decoder 1301, the sound-source parameter decoder 1302, the adaptation sign book 1303, a multiplier 1304, the noise sign book 1305 that consists of a partial algebraic-sign book and a random sign book, a multiplier 1306, an adder 1307, the LPC composition filter 1308, and the after-treatment machine 1309.
[0141] In this voice decryption equipment, LPC coded data and sound-source parameter coded data are inputted into the LPC decoder 1301 and the sound-source parameter decoder 1302 per frame through a transmission line, respectively. The LPC decoder 1301 decodes Quantization LPC and outputs it to the LPC composition filter 1308. When using Quantization LPC with the after-treatment vessel 1309, Quantization LPC is outputted also to the after-treatment machine 1309 from the LPC decoder 1301 at coincidence. The sound-source parameter decoder 1302 outputs the positional information which starts an adaptation sign vector, adaptation sign vector gain, the index information which specifies a noise sign vector, and noise sign vector gain, respectively to the noise sign book 1305 which consists of the adaptation sign book 1303, a multiplier 1304, and a partial algebraic-sign book and a random sign book, and a multiplier 1306.
[0142] The adaptation sign book 1303 is the buffer of the excitation vector (vector outputted from an adder 1307) generated in the past, starts an adaptation sign vector from the logging location inputted from the sound-source parameter decoder 1302, and outputs it to a multiplier 1304. A multiplier 1304 multiplies the adaptation sign vector outputted from the adaptation sign book 1303 by the adaptation sign vector gain inputted from the sound-source parameter decoder 1302, and outputs it to an adder 1307.
[0143] The noise sign book 1305 which consists of a partial algebraic-sign book and a random sign book It is a noise sign book with the configuration shown in drawing 14 , and is the same noise sign book as what was shown in 1206 of drawing 12 . Either the noise sign vector which consists of the pulse which is [several / the location of at least two pulses specified by the index inputted from the sound-source parameter decoder 1302 is close], or the noise sign vector of about 90% or less of rates of sparse is outputted to a multiplier 1306.
[0144] A multiplier 1306 multiplies the noise sign vector outputted from the partial algebraic-sign book by the noise sign vector gain inputted from the sound-source parameter decoder 1302, and outputs it to an adder 1306. By performing vector addition with the adaptation sign vector after the adaptation sign vector gain multiplication outputted from a multiplier 1304, and the noise sign vector after the noise sign vector gain multiplication outputted from the multiplier 1306, an adder 1307 generates an excitation vector and outputs it to the adaptation sign book 1303 and the LPC composition filter 1308.
[0145] The excitation vector outputted to the adaptation sign book 1303 is used when updating the adaptation sign book 1303, and the excitation vector outputted to the LPC composition filter 1308 is used in order to generate synthesized speech. The LPC composition filter 1308 is a linear prediction filter constituted using the quantization LPC outputted from the LPC decoder 1301, drives an LPC composition filter using the excitation vector outputted from the adder 1307, and outputs a composite signal to the after-treatment machine 1309.
[0146] The processing for improving subjective quality, such as processing for making easy to hear to the synthesized speech outputted from the LPC composition filter 1308 the postfilter processing and the steady background noise which consist of formant emphasis processing, pitch emphasis processing, spectrum inclination amendment processing, etc., is carried out, and the after-treatment machine 1309 outputs as decode voice data.
[0147] The configuration of the noise sign vector generation equipment concerning the gestalt 2 of the operation of this invention to drawing 14 is shown. The noise sign vector generation equipment shown in this drawing is equipped with the partial algebraic-sign book 1401 and the random sign book 1402 which were shown in the gestalt 1 of operation.
[0148] The partial algebraic-sign book 1401 generates the noise sign vector to which at least two pulses which consist of two or more unit pulses are close, and outputs it to a circuit changing switch 1403. The generation method of the noise sign vector of the partial algebraic-sign book 1401 is concretely shown in the gestalt 1 of operation.
[0149] The random sign book 1402 chooses one vector from the noise sign vectors which store the noise sign vector which consists of many pulse numbers, and are stored rather than the noise sign vector generated from the partial algebraic-sign book 1401, and outputs it to a change-over switch 1403.
[0150] The random sign book 1402 is [using a sign book with more independent constituting from two or more channels rather than] advantageous in respect of the amount of operations, and the amount of memory. Moreover, since a noise sign vector which two pulses are approaching is generable by the partial algebraic-sign book 1401, it can improve the engine performance to a voiceless consonant or stationary noise by storing in the random sign book 1402 the noise sign vector the pulse stands on ** etc. at the whole frame which no pulses are approaching.
[0151] Moreover, in order to lessen the amount of operations by the case where frame length is 80 samples, as for the

pulse number of the noise sign vector which the random sign book 1401 stores, it is desirable to carry out to about 8-16. In this case, what is necessary is just to store the vector which consists of the pulse of each about 4-8 channels, if the random sign book 1401 is made a two-channel configuration. Moreover, it is also possible by setting the amplitude of each pulse to +1 or -1 in such a sparse vector to aim at saving of the amount of operations and the amount of memory further.

[0152] A change-over switch 1403 is control (for example, control is received from the block which performs error minimization with a target when using this noise sign vector for an encoder) from the outside. One of the noise sign vector outputted from the partial algebraic-sign book 1401 controlled by the index of the noise sign vector decoded when using for a decryption machine, and the noise sign vectors outputted from the random sign book 1402 is chosen. It outputs as an output noise sign vector 1404 of a noise sign vector generation machine.

[0153] Here, it is desirable that they are [of the noise sign vector outputted from the random sign book 1402 and the noise sign vector outputted from the partial algebraic-sign book 1401] 1:1-2:1 [50 - 66%], i.e., random, and 34 - 50% of algebra comparatively (random: algebra).

[0154] With reference to drawing 15 , it explains below that processing of the noise sign vector generation method (the coding approach, the noise sign book retrieval approach) in the gestalt of the above-mentioned implementation flows. First, it looks for a partial algebraic-sign book in ST1501. As the detail of the concrete retrieval approach is shown in the gestalt 1 of operation, it realizes by maximizing a formula (1). The size of a partial algebraic-code book is IDXa, and the index index of the optimal candidate out of a partial algebraic-sign book (0 <=index<IDXa) is determined at this step.

[0155] Next, it looks for a random sign book in ST1502. Retrieval of a random sign book is performed using the approach which is generally performed and shines with a CELP encoder. The valuation plan shown in a formula (1) is specifically calculated to all the noise sign vectors in which it is stored by the random sign book, and the index index to the vector used as max is determined. however, since maximization of a formula (1) is performed in ST1501, only when the noise sign vector exceeding the maximum of the formula (1) determined by ST1501 exists, index determined by ST1501 is already updated on the new index index (IDXa<=index -- < (IDXa+IDXr)). When the noise sign vector exceeding the maximum of the formula (1) determined by ST1501 is not stored in the random sign book, the coded data (index index) determined by ST1501 is outputted as encoded information of a noise sign vector.

[0156] With reference to drawing 16 , the flow of processing of the noise sign vector generation method (the decryption approach) in the gestalt of the above-mentioned implementation is explained below.

[0157] The encoded information index of the noise sign vector first transmitted and decoded from the encoder in ST1601 judges whether it is under IDXa. IDXa is the size of a partial noise sign book. This noise sign vector generation machine is generating the noise sign vector from the noise sign book which consists of the partial algebraic-sign book of size IDXa, and the random sign book of size IDXr, and, as for this noise sign book, the index is equipped with the IDXa and random - (IDXa+IDXr -1) sign book for the 0 and partial - (IDXa-1) algebraic-sign book. Therefore, with [received index] IDXa [under], a noise sign vector is generated by the partial algebraic-sign book, and with IDXa [more than] (IDXa+IDXr) (following), a noise sign vector will be generated by the random sign book. With [in this step / index] IDXa [under], it progresses to ST1602, and with IDXa [more than], it progresses to ST1604.

[0158] In ST1602, decode of a partial algebraic-sign book parameter is performed. The concrete decode approach is shown in the gestalt 1 of operation. For example, when the number of pulses is two, the location position1 of the 1st pulse and the location position2 of the 2nd pulse are decoded from Index index. Moreover, when the polar information on a pulse is also included in index, the polarity sign 1 of the 1st pulse and the polarity sign 2 of the 2nd pulse are decoded collectively. sign1 and sign2 are +1 or -1 here.

[0159] Next, in ST1603, a noise sign vector is generated from the decoded partial algebraic-sign book parameter. When the number of pulses is two, the amplitude specifically stands [a polarity] the pulse of 1 by sign1 at the location of position1, the amplitude stands [a polarity] the pulse of 1 by sign2 at the location of position2, and all the other points output the vector code [0-Num-1] set to 0 as a noise sign vector. Here, Num is frame length or noise sign vector length (sample).

[0160] On the other hand, in ST1601, when index is more than IDXa, it progresses to ST1604. In ST1604, IDXa is subtracted from index. This is for only changing index into the range of 0-IDXr-1. IDXr is the size of a random sign book here.

[0161] Next, decode of a random sign book parameter is performed in ST1605. Specifically, in the case of the random sign book of a two-channel configuration, the random sign book index indexR1 of the 1st channel and the random sign book index indexR2 of the 2nd channel are decoded from index. Moreover, when the polar information on each

channel is included in index, the polarity sign 1 of the 1st channel and the polarity sign 2 of the 2nd channel are decoded collectively. sign1 and sign2 are +1 or 1.
[0162] Next, in ST1606, a noise sign vector is generated from the decoded random sign book parameter. When a random sign book is a two-channel configuration, RCB2 from 2nd channel RCB2 [indexR2] and [0-Num-1] are specifically taken out for RCB1 from 1st channel RCB1 [indexR1], and [0-Num-1], respectively, and the thing adding two vectors is outputted as a noise sign vector code [0-Num-1]. Here, Num is frame length or noise sign vector length (sample).
[0163] (Gestalt 3 of operation) Drawing 17 is the block diagram having shown the voice to digital converter equipped with the noise sign vector generation machine concerning the gestalt 3 of operation. The voice to digital converter shown in this drawing is equipped with the pretreatment machine 1701, the LPC analyzer 1702, the LPC quantizer 1703, the adaptation sign book 1704, a multiplier 1705, the noise sign book 1706 that consists of a partial algebraic-sign book and a random sign book, a multiplier 1707, an adder 1708, the LPC composition filter 1709, an adder 1710, the acoustic-sense weighting machine 1711, the error minimization machine 1712, and the mode judging machine 1713.
[0164] In this voice to digital converter, input voice data is a digital signal acquired by carrying out A/D conversion of the sound signal, and is inputted into the pretreatment machine 1701 at every batch time amount (frame). The pretreatment machine 1701 improves input voice data in quality subjectively, or performs processing for changing into the signal in the condition of having been suitable for coding, and performs pre-emphasis processing which emphasizes the description of the high-pass filter processing for cutting a dc component for example, or a sound signal.
[0165] The signal after pretreatment is outputted to the LPC analyzer 1702 and an adder 1710. The LPC analyzer 1702 performs LPC analysis (linear predictive coding) using the signal inputted from the pretreatment machine 1701, and outputs obtained LPC (linear predictor coefficients) to the LPC quantizer 1703. The LPC quantizer 904 quantizes LPC inputted from the LPC analyzer 903, outputs Quantization LPC to the LPC composition filter 1709 and the mode judging machine 1713, and outputs the coded data of Quantization LPC to a decoder side through a transmission line.
[0166] The mode judging machine 1713 outputs to dynamic and the noise sign book 1716 which uses the static description, performs carving between the voice section, the non-voice section or the voiced section, and non-vocal register (mode judging), and consists a judgment result of a partial algebraic-sign book and a random sign book of the inputted quantization LPC. By using the dynamic description of Quantization LPC, the voice section / non-voice section is carved, and, more specifically, it performs carving between voiced / non-vocal register by using the static description of Quantization LPC. The distance (difference) of the average quantization LPC in the section judged as a dynamic description of Quantization LPC in the inter-frame amount of fluctuation and the inter-frame past to be the non-voice section and the quantization LPC in the present frame etc. can be used. Moreover, the primary reflection coefficient etc. can be used as a static description of Quantization LPC.
[0167] In addition, Quantization LPC can be more effectively used by changing into the parameter of other fields, such as LSP, a reflection coefficient, and LPC prediction remainder PAWA. Moreover, when it is possible to transmit mode information, a mode judging cannot be performed from Quantization LPC, but a more exact and fine mode judging can also be performed using various parameters which analyze input voice data and are obtained. In this case, it encodes and mode information is outputted to a decoder side through a transmission line with the LPC coded data 1714 and the sound-source parameter coded data 1715.
[0168] The adaptation sign book 1704 is the buffer of the excitation vector (vector outputted from an adder 1708) generated in the past, starts an adaptation sign vector from the location specified with the error minimization vessel 1712, and outputs it to a multiplier 1705. A multiplier 1705 multiplies the adaptation sign vector outputted from the adaptation sign book 1704 by adaptation sign vector gain, and outputs it to an adder 1708.
[0169] Adaptation sign vector gain is specified with an error minimization vessel. The noise sign book 1706 which consists of a partial algebraic-sign book and a random sign book As it is the noise sign book from which the ratio of a partial algebraic-sign book and a random sign book changes using the mode information inputted from the mode judging machine 1713 and is shown in drawing 9 It has the configuration with which the number of entries of a partial algebraic-sign book and the number of entries of a random sign book are controlled by mode information accommodative (switched). Either the noise sign vector which consists of the pulse which is [several / the location of at least two pulses is close], or the noise sign vector of about 90% or less of rates of sparse (the measurement size of the amplitude zero to the measurement size of the whole frame comparatively) is outputted to a multiplier 1707.
[0170] A multiplier 1707 multiplies the noise sign vector outputted from the noise sign book 1706 which consists of a partial algebraic-sign book and a random sign book by noise sign vector gain, and outputs it to an adder 1708. By

performing vector addition with the adaptation sign vector after the adaptation sign vector gain multiplication outputted from the multiplier 1705, and the noise sign vector after the noise sign vector gain multiplication outputted from the multiplier 1707, an adder 1708 generates an excitation vector and outputs it to the adaptation sign book 1704 and the LPC composition filter 1709.

[0171] The excitation vector outputted to the adaptation sign book 1704 is used for updating the adaptation sign book 1704, and the excitation vector outputted to the LPC composition filter 1709 is used in order to generate synthesized speech. The LPC composition filter 1709 is a linear prediction filter constituted using the quantization LPC outputted from the LPC quantizer 1703, drives an LPC composition filter using the excitation vector outputted from the adder 1708, and outputs a composite signal to an adder 1710.

[0172] the difference (error) of the input sound signal after pretreatment to which the adder 1710 was outputted from the pretreatment machine 1701, and the composite signal outputted from the LPC composition filter 1709 -- a signal is calculated and it outputs to the acoustic-sense weighting machine 1711. The acoustic-sense weighting machine 1711 performs acoustic-sense-weighting by considering the differential signal outputted from an adder 1710 as an input, and outputs it to the error minimization machine 1712.

[0173] The error minimization machine 1712 considers the differential signal after acoustic-sense weighting outputted from the acoustic-sense weighting machine 1711 as an input. For example, the noise sign vector generated from the noise sign book 1706 which consists of the location and partial algebraic-sign book which start an adaptation sign vector from the adaptation sign book 1704, and a random sign book so that the square sum may serve as min, The value of the adaptation sign vector gain by which it multiplies with a multiplier 1705, and the noise sign vector gain by which it multiplies with a multiplier 1707 is adjusted, each is encoded, and it outputs to a decoder side through a transmission line as sound-source parameter coded data.

[0174] Drawing 18 shows voice decryption equipment equipped with the noise sign vector generation machine concerning the gestalt 3 of operation. The voice decryption equipment shown in this drawing is equipped with the LPC decoder 1801, the sound-source parameter decoder 1802, the adaptation sign book 1803, a multiplier 1804, the noise sign book 1805 that consists of a partial algebraic-sign book and a random sign book, a multiplier 1806, an adder 1807, the LPC composition filter 1808, the after-treatment machine 1809, and the mode judging machine 1810.

[0175] In this voice decryption equipment, LPC coded data and sound-source parameter coded data are inputted into the LPC decoder 1801 and the sound-source parameter decoder 1802 per frame through a transmission line, respectively. The LPC decoder 1801 decodes Quantization LPC and outputs it to the LPC composition filter 1808 and the mode judging machine 1810. When using Quantization LPC with the after-treatment vessel 1809, Quantization LPC is outputted also to the after-treatment machine 1809 from the LPC decoder 1801 at coincidence. The mode judging machine 1810 is the same configuration as the mode judging machine 1713 of drawing 17 , and outputs to dynamic, and the noise sign book 1805 and the after-treatment machine 1809 which uses the static description, performs carving between the voice section, the non-voice section or the voiced section, and non-vocal register (mode judging), and consists a judgment result of a partial algebraic-sign book and a random sign book of the inputted quantization LPC.

[0176] By using the dynamic description of Quantization LPC, the voice section / non-voice section is carved, and, more specifically, it performs carving between voiced / non-vocal register by using the static description of Quantization LPC. The distance (difference) of the average quantization LPC in the section judged as a dynamic description of Quantization LPC in the inter-frame amount of fluctuation and the inter-frame past to be the non-voice section and the quantization LPC in the present frame etc. can be used. Moreover, the primary reflection coefficient etc. can be used as a static description of Quantization LPC.

[0177] In addition, Quantization LPC can be used more effectively by changing into the parameter of other fields, such as LSP, a reflection coefficient, and LPC prediction remainder PAWA. Moreover, when it is possible to transmit mode information as another information, the mode information transmitted separately is decoded and decode mode information is outputted to the noise sign book 1805 and the after-treatment machine 1809.

[0178] The sound-source parameter decoder 1802 outputs the positional information which starts an adaptation sign vector, adaptation sign vector gain, the index information which specifies a noise sign vector, and noise sign vector gain, respectively to the noise sign book 1805 which consists of the adaptation sign book 1803, a multiplier 1804, a partial algebraic-sign book, and a random sign book, and a multiplier 1806.

[0179] The adaptation sign book 1803 is the buffer of the excitation vector (vector outputted from an adder 1807) generated in the past, starts an adaptation sign vector from the logging location inputted from the sound-source parameter decoder 1802, and outputs it to a multiplier 1804. A multiplier 1804 multiplies the adaptation sign vector outputted from the adaptation sign book 1803 by the adaptation sign vector gain inputted from the sound-source

parameter decoder 1802, and outputs it to an adder 1807.

[0180] The noise sign book 1807 which consists of a partial algebraic-sign book and a random sign book It is a noise sign book with the configuration shown in drawing 9 , and is the same noise sign book as what was shown in 1706 of drawing 17 . Either the noise sign vector which consists of the pulse which is [several / the location of at least two pulses specified by the index inputted from the sound-source parameter decoder 1802 is close], or the noise sign vector of about 90% or less of rates of sparse is outputted to a multiplier 1806.

[0181] A multiplier 1806 multiplies the noise sign vector outputted from the partial algebraic-sign book by the noise sign vector gain inputted from the sound-source parameter decoder 1802, and outputs it to an adder 1806. By performing vector addition with the adaptation sign vector after the adaptation sign vector gain multiplication outputted from a multiplier 1804, and the noise sign vector after the noise sign vector gain multiplication outputted from the multiplier 1806, an adder 1807 generates an excitation vector and outputs it to the adaptation sign book 1803 and the LPC composition filter 1808.

[0182] The excitation vector outputted to the adaptation sign book 1803 is used for updating the adaptation sign book 1803, and the excitation vector outputted to the LPC composition filter 1808 is used in order to generate synthesized speech. The LPC composition filter 1808 is a linear prediction filter constituted using the quantization LPC outputted from the LPC decoder 1801, drives an LPC composition filter using the excitation vector outputted from the adder 1807, and outputs a composite signal to the after-treatment machine 1809.

[0183] The processing for improving subjective quality, such as processing for making easy to hear to the synthesized speech outputted from the LPC composition filter 1808 the postfilter processing and the steady background noise which consist of formant emphasis processing, pitch emphasis processing, spectrum inclination amendment processing, etc., is carried out, and the after-treatment machine 1809 outputs as decode voice data 1810. Such after treatment is performed accommodative using the mode information inputted from the mode judging machine 1808. That is, the after treatment for which it was suitable for every mode is changed and applied, or the strength of after treatment is changed accommodative.

[0184] Drawing 19 is the block diagram showing the configuration of the noise sign vector generation equipment concerning the gestalt 3 of operation of this invention. The noise sign vector generation machine shown in this drawing is equipped with the pulse-position limited machine controller 1901, the partial algebraic-sign book 1902, the number controller 1903 of random sign book entries, and the random sign book 1904.

[0185] The pulse-position limited machine controller 1901 outputs the control signal of a pulse-position limited machine to the partial algebraic-sign book 1902 according to the mode information inputted from the outside. Size of a partial algebraic-sign book is made small by the thing which this control makes fluctuate size of a partial algebraic-sign book (responding to the mode) and for which it carries out for accumulating, and limitation is strengthened case [whose mode is / like silent / stationary noise mode] (the number of candidates of the pulse position is lessened) (instead, it controls by the number controller 1903 of random sign book entries so that the size of the random sign book 1904 becomes large).

[0186] By doing in this way, if the noise sign vector which consists of several pulses, such as the silent section and the stationary noise section, is used, it will become possible to aim at an engine-performance improvement to a signal with which subjective quality deteriorates. The pulse-position limited machine is built into the partial algebraic-sign book 1902, and the concrete actuation is shown in the gestalt 1 of operation.

[0187] The partial algebraic-sign book 1902 is a partial algebraic-sign book by which actuation of the pulse-position limited machine built into the interior is controlled by the control signal inputted from the pulse-position limited machine controller 1901, and sign book size fluctuates it by the limited degree of a pulse-position candidate with a pulse-position limited machine. Concrete actuation of a partial algebraic-sign book is shown in the gestalt 1 of operation. The noise sign vector generated from this sign book is outputted to a change-over switch 1905.

[0188] The number controller 1903 of random sign book entries performs control which fluctuates the size of the random sign book 1904 according to the mode information inputted from the outside. This control is performed by control of the pulse-position limited machine controller 1901 being interlocked with. That is, when the number controller 1903 of random sign book entries decreases the size of the random sign book 1904 when the size of the partial algebraic-sign book 1902 is made to increase with the pulse-position limited machine controller 1901, and decreasing the size of the partial algebraic-sign book 1902 with the pulse-position limited machine controller 1901, the number controller 1903 of random sign book entries performs control to which the size of the random sign book 1904 is made to increase. And the total number of entries (total sign book size in this noise sign vector generation machine) which set the partial algebraic-sign book 1902 and the random sign book 1904 is always maintained at a fixed value.

[0189] The random sign book 1904 generates a noise sign vector using the random sign book of the size which inputted the control signal from the number controller 1903 of random sign book entries, and was specified, and outputs it to a change-over switch 1905. It is more effective from the field of the amount of memory to use it as a random sign book of two or more sizes by consisting of only random sign books of one kind to share of existing size which was defined, and using this partially, although the random sign book 1904 may consist of random sign books of size by which plurality differs here.
[0190] Moreover, although a sign book independent one channel is sufficient as the random sign book 1904, it is more advantageous from the field of the amount of operations, or the amount of memory to use the sign book which consists of two or more two or more channels.
[0191] a change-over switch 1905 -- the control (the control signal from the block which minimizes an error with a target vector when using this noise sign vector generation machine for an encoder --) from the outside Using the parameter information on the noise sign book decoded when using for a decryption machine etc. One of the noise sign vectors outputted from the partial algebraic-sign book 1902 or the random sign book 1904 is chosen, and it outputs as an output noise sign vector 1906 of this noise sign vector generation machine.
[0192] Here, it is [in / comparatively (random: algebra) / voiced mode] desirable that they are [of the noise sign vector outputted from the random sign book 1904 and the noise sign vector outputted from the partial algebraic-sign book 1902] 0:1-1:2 [0 - 34%], i.e., random, and 66 - 100% of algebra. Moreover, as for the above-mentioned rate (random: algebra), in non-voiced mode, it is desirable that they are 2:1-4:1 [66 - 80%], i.e., random, and 20 - 34% of algebra.
[0193] With reference to drawing 20 , it explains below that processing of the noise sign vector generation method (the coding approach) in the gestalt of the above-mentioned implementation flows.
[0194] First, in ST2001, size of a partial algebraic-sign book and a random sign book is set up based on the mode information inputted separately. At this time, a setup of the size of a partial algebraic-sign book is performed by fluctuating the number of location candidates of the pulse which is shown in the gestalt 1 of operation and by which a relative-position expression is carried out.
[0195] This change in a pulse by which a relative-position expression is carried out can be performed mechanically, and it is made to decrease by reducing from the part which a relative position leaves. When relative positions are {1, 3, 5, 7}, more specifically, the number of location candidates is reduced like {{1, 3, 5}, {1, 3}, 1}. Conversely, like {1} to {1, 3}, and {1, 3, 5}, when increasing, it increases.
[0196] Moreover, a setup of the size of a partial algebraic-sign book and a random sign book is performed so that total of the size of a partial algebraic-sign book and a random sign book may become constant value. In the mode which corresponds to the voiced (stationary) section, the size (ratio) of a partial algebraic-sign book is more specifically large, and size of both the sign book is set up so that the size (ratio) of a random sign book may become large in the mode which corresponds to the silent section or the noise section.
[0197] the mode information and IDXa which mode inputted in this block -- the size (the number of noise sign vector entries) of a partial algebraic-sign book, and IDXr -- random sign book size (the number of noise sign vector entries) -- it is -- IDXa+IDXr= constant value -- it comes out. Moreover, a setup of the number of entries of a random sign book is realizable by setting up the range of the random sign book referred to, for example. For example, in control which switches and uses the size of the random sign book of two channels by 128x128=16384 and 64x64=4096, it is easily realizable by switching the range of an index which it has the random (indexes 0-127) sign book which stores the vector of 128 kinds of each channel, respectively, and is searched for it by two kinds, 0-127, and 0-63.
[0198] In addition, as for the vector space where the vector of indexes 0-127 exists in this case, and the vector space where the vector of indexes 0-63 exists, it is desirable that it is in agreement as much as possible. If the vector of indexes 64-127 cannot be expressed at all, namely, the vector space of indexes 0-63 completely differs from the vector space of indexes 64-127 by the vector of indexes 0-63 Since modification of the above random sign book sizes may degrade the coding engine performance of a random sign book greatly, it needs to create a random sign book in consideration of such a thing.
[0199] In addition, inevitably, when keeping constant total of the number of entries of a partial algebraic-sign book and a random sign book, since the method (put together) of sizing of both the sign book is limited to some kinds, it serves as ** [switch / control of sizing / a setup of these some kinds]. In Book ST, the partial algebraic-sign book size IDXa and the random sign book size IDXr are set up from the inputted mode information mode.
[0200] Next, in ST2002, the noise sign vector which makes an error with a target vector the smallest is chosen from a partial algebraic-sign book (size IDXa) and a random sign book (IDXr), and it asks for the index. It is determined that it will become the range of - (IDXa+IDXr -1) if a noise sign vector is chosen for example, from a partial algebraic-

sign book and Index index will be chosen from a 0 - (IDXa-1) random sign book (IDXa-1).

[0201] Next, in ST2003, the called-for index index is outputted as coded data. index is encoded by the form further outputted to a transmission line if needed.

[0202] With reference to drawing 21 , it explains below that processing of the noise sign vector generation method (the decryption approach) in the gestalt of the above-mentioned implementation flows.

[0203] First, in ST2101, size of a partial algebraic-sign book and a random sign book is set up based on the mode information mode decoded separately. The approach of a concrete setup is as the above-mentioned explained with reference to drawing 20 . The size IDXa of a partial algebraic-sign book and the size IDXr of a random sign book are set up from the mode information mode.

[0204] Next, in ST2102, a noise sign vector is decoded using a partial algebraic-sign book or a random sign book. it determines using which sign book it decodes by the value of the index index of the noise sign vector decoded separately -- having -- the case of 0 <=index<IDXa -- IDXa<=index from a partial algebraic-sign book -- < (IDXa+IDXr) -- a case is decoded from a random sign book. With the gestalt 3 of operation, with reference to drawing 16 , as it explained, specifically, it decodes.

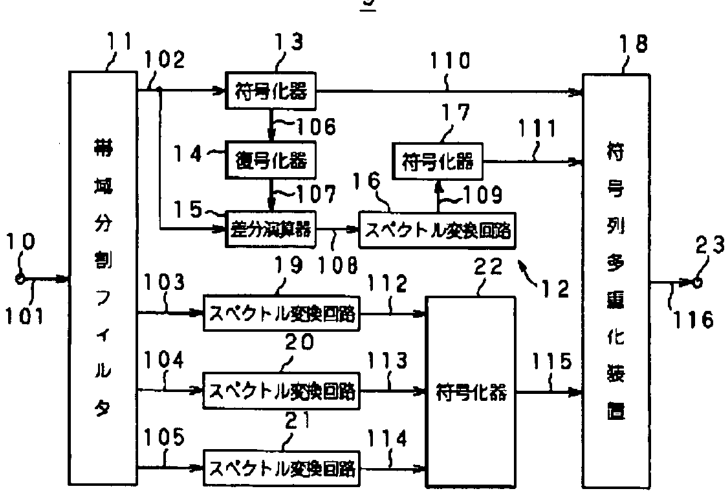
[0205] In addition, if the above indexes are given, a different index to the entry of the noise sign vector shared in the different mode will be given. Since it becomes easy to be influenced of [when (that is, it becoming an index which is different when the modes' differ also by the noise sign vector which has the completely same configuration), and a transmission-line error arise]

* NOTICES *

1.This document has been translated by computer. So the translation may not reflect the original precisely.
2.**** shows the word which can not be translated.
3.In the drawings, any words are not translated.

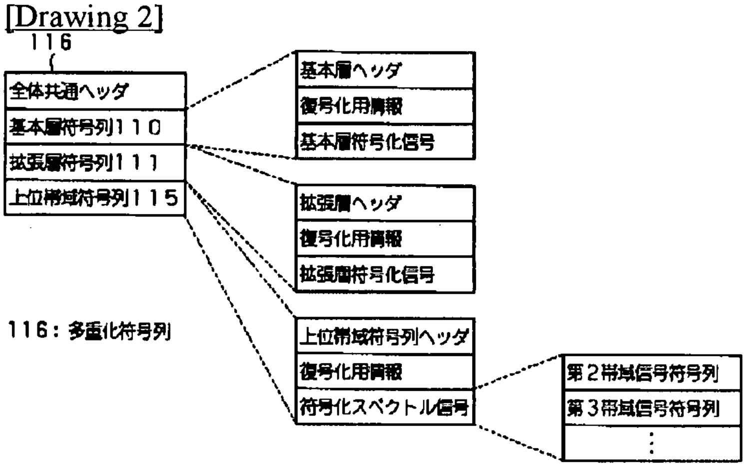
DRAWINGS

[Drawing 1]



9：符号化装置
12：階層符号化部

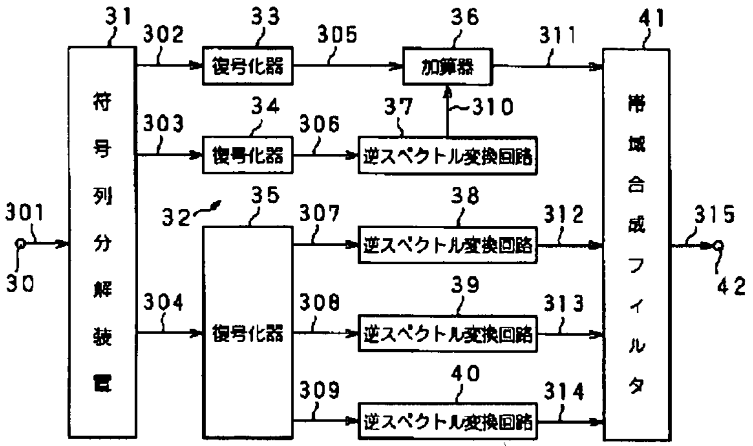
[Drawing 2]



116：多重化符号列

[Drawing 3]

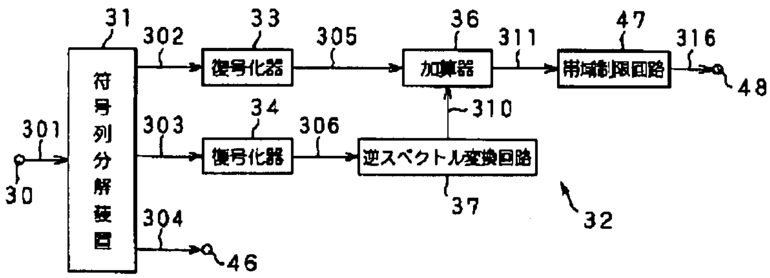
29



29：復号化装置
32：階層復号化部

[Drawing 4]

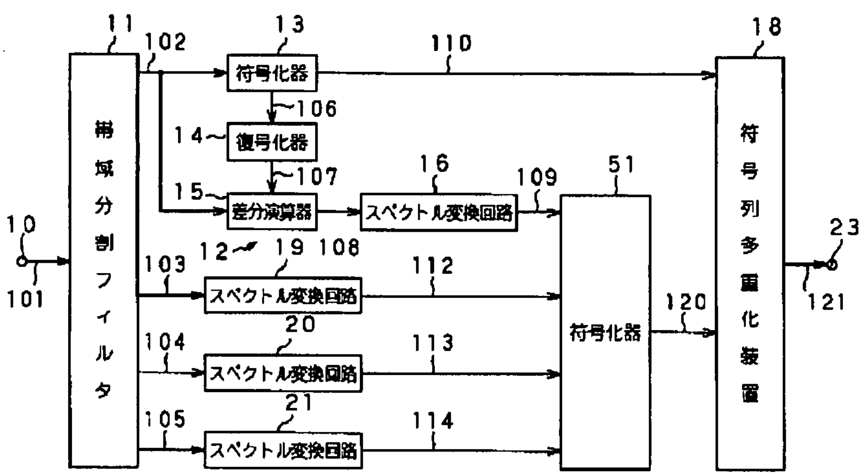
45



32：階層復号化部
45：復号化装置

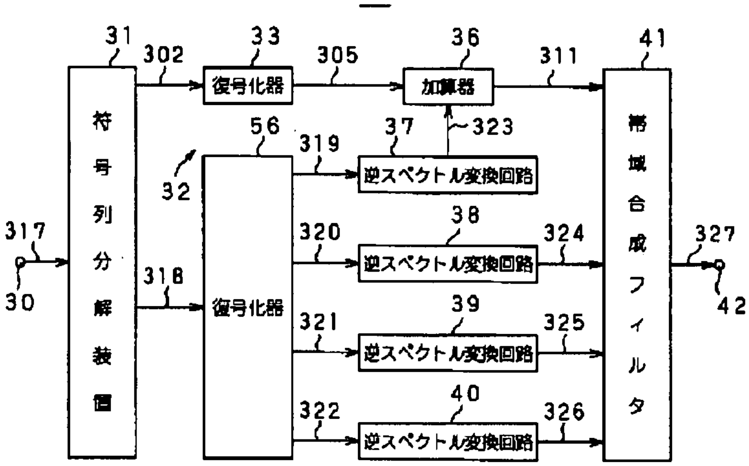
[Drawing 5]

50



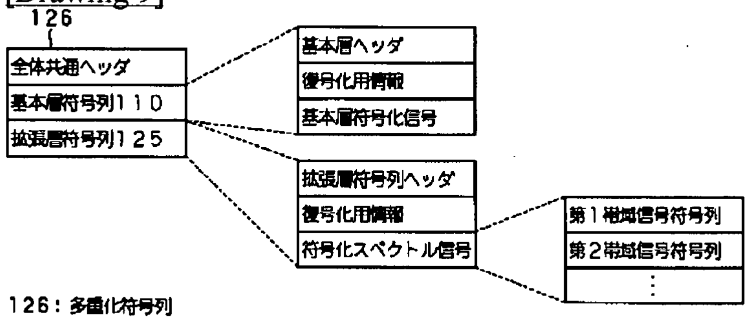
50：符号化装置
12：階層符号化部

[Drawing 6]



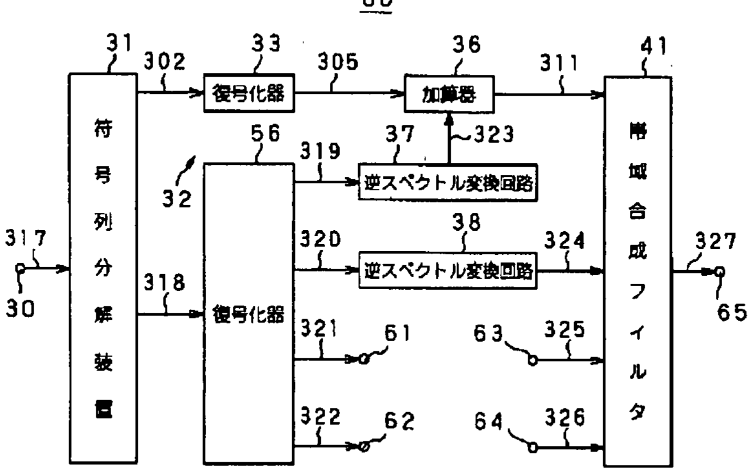
55：復号化装置
32：階層復号化部

[Drawing 9]



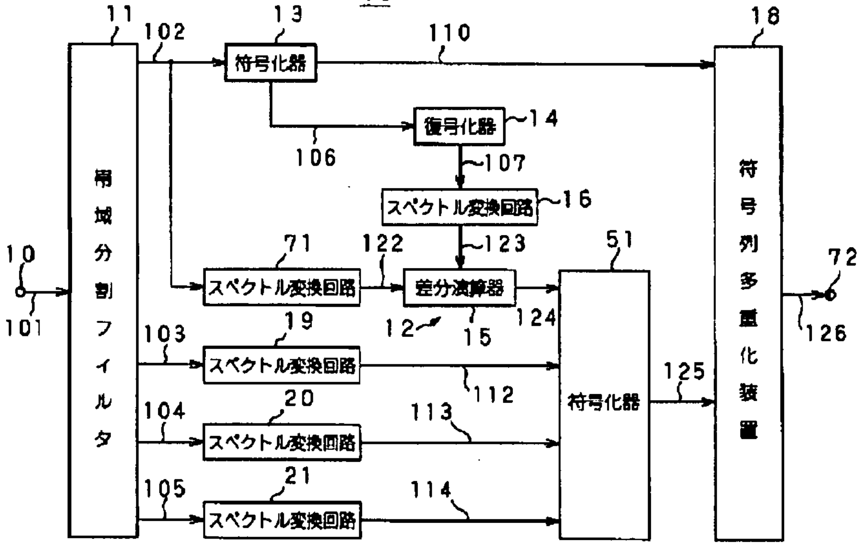
126：多重化符号列

[Drawing 7]



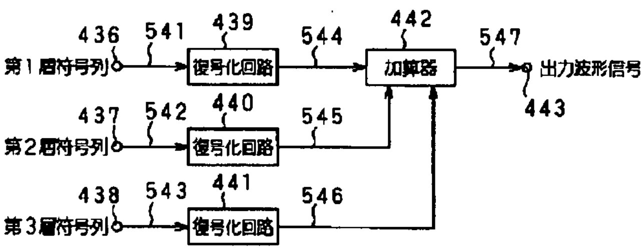
60：復号化装置
32：階層復号化部

[Drawing 8]



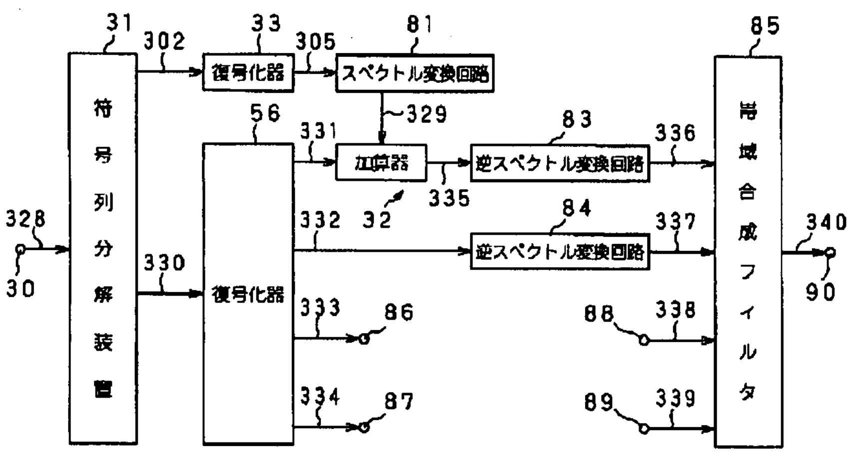
70 符号化装置
12：階層符号化部

[Drawing 16]



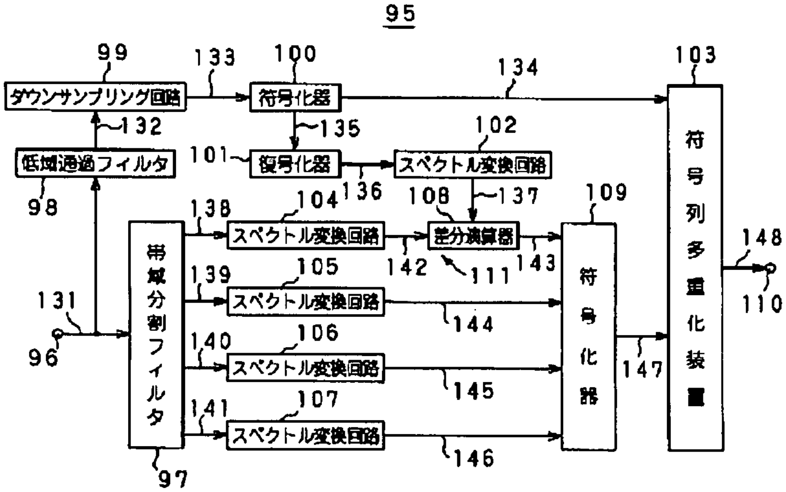
435：復号化装置

[Drawing 10]



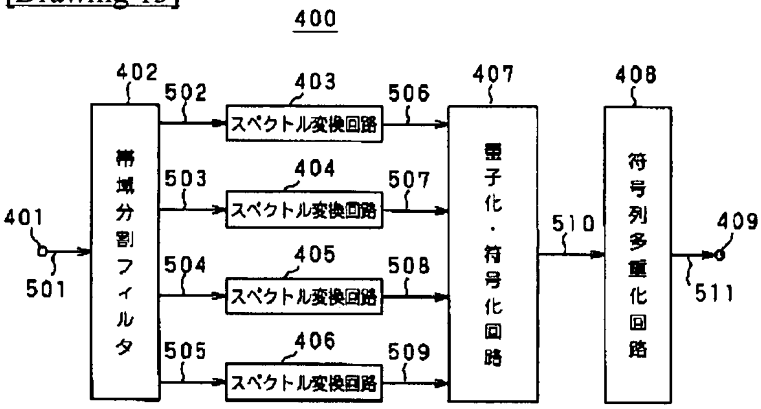
80：復号化装置
32：階層復号化部

[Drawing 11]



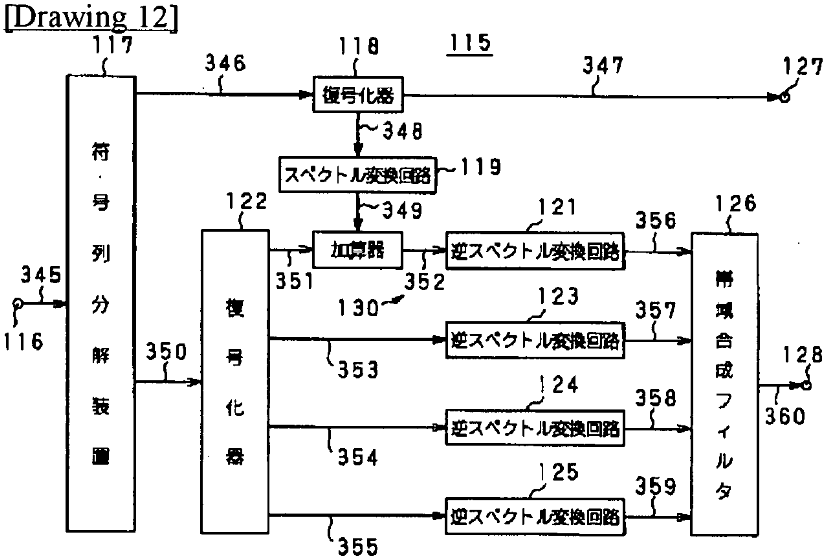
95：符号化装置
111：階層符号化部

[Drawing 13]



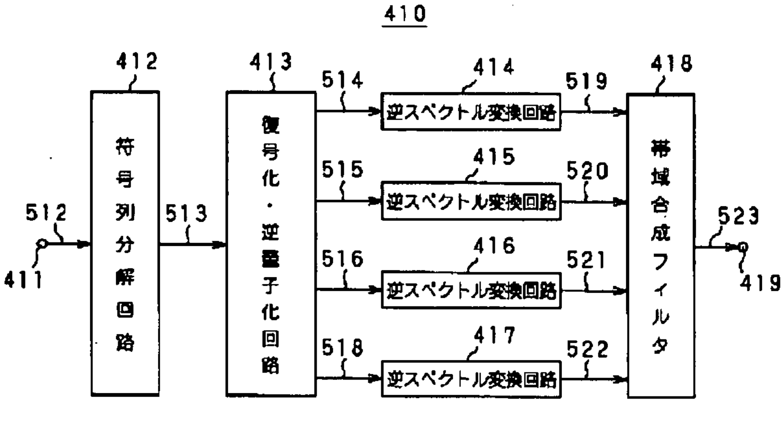
400：符号化装置

[Drawing 12]



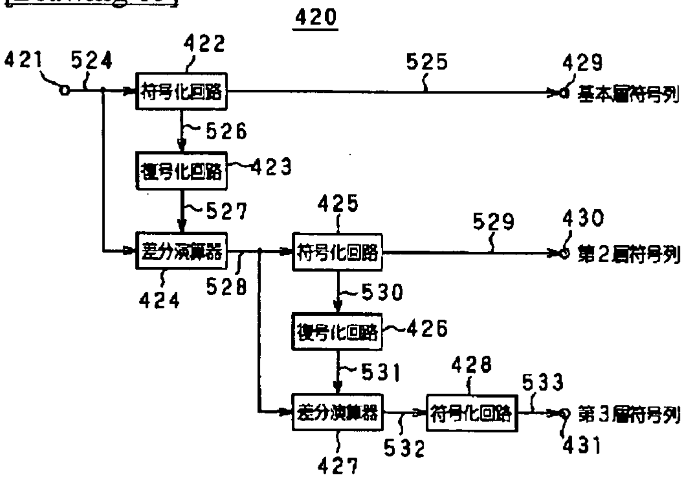
115：復号化装置
130：階層復号化部

[Drawing 14]



410：復号化装置

[Drawing 15]



420：符号化装置

[Translation done.]